US010134403B2

(12) **United States Patent**
Kim et al.

(10) **Patent No.: US 10,134,403 B2**
(45) **Date of Patent: Nov. 20, 2018**

(54) **CROSSFADING BETWEEN HIGHER ORDER AMBISONIC SIGNALS**

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(73) Assignee: **QUALCOMM Incorporated**, San Diego, CA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **14/712,854**

(22) Filed: **May 14, 2015**

(65) **Prior Publication Data**

US 2015/0332683 A1 Nov. 19, 2015

Related U.S. Application Data

(60) Provisional application No. 61/994,763, filed on May 16, 2014, provisional application No. 62/004,076, filed on May 28, 2014, provisional application No. 62/118,434, filed on Feb. 19, 2015.

(51) **Int. Cl.**
G10L 19/008 (2013.01)

(52) **U.S. Cl.**
CPC ***G10L 19/008*** (2013.01)

(58) **Field of Classification Search**
CPC G10L 19/008
USPC 705/500
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

2005/0157894	A1*	7/2005	Andrews	H04S 7/30 381/307
2012/0053710	A1*	3/2012	Lindahl	G11B 20/10 700/94
2014/0016786	A1*	1/2014	Sen	G10L 19/008 381/23
2014/0355769	A1	12/2014	Peters et al.	
2015/0213803	A1	7/2015	Peters et al.	
2016/0057556	A1*	2/2016	Boehm	G10L 21/02 381/23

FOREIGN PATENT DOCUMENTS

WO 2014194099 A1 12/2014

OTHER PUBLICATIONS

Boehm, et al., "Scalable Decoding Mode for MPEG-H 3D Audio HOA," MPEG Meeting; Mar. 2014; Valencia; (Motion Picture Expert Group or ISO/IEC JTC1/SC29/WG11), No. m33195, XP030061647, 12 pp.

(Continued)

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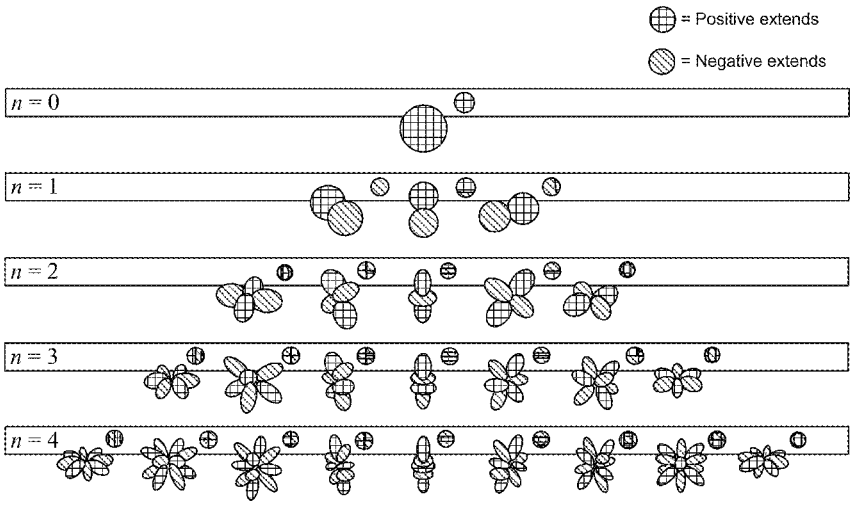
(57) **ABSTRACT**

In general, techniques are described for crossfading sets of spherical harmonic coefficients. An audio encoding device or audio decoding device comprising a memory and a processor may be configured to perform the techniques. The memory may be configured to store a first set of spherical harmonic coefficients (SHCs) and a second set of SHCs. The first set of SHCs describe a first sound field. The second set of SHCs describe a second sound field. The processor may be configured to crossfade between the first set of SHCs and a second set of SHCs to obtain a first set of crossfaded SHCs.

26 Claims, 14 Drawing Sheets



(56) **References Cited**

OTHER PUBLICATIONS

Peters, et al., "Description of Qualcomm's HoA coding technology", MPEG Meeting; Jul. 2013; Vienna; (Motion Picture Expert Group or ISO/IEC JTC1/SC29/WG11), No. m29986, XP030058515, 3 pp.

Sen, et al., "Technical Description of the Qualcomm's HoA Coding Technology for Phase II", MPEG Meeting; Jul. 2014; Sapporo; (Motion Picture Expert Group or ISO/IEC JTC1/SC29/WG11), No. m34104, XP030062477, 4 pp.

Sen, et al., "Thoughts on scalable/layered coding for the HOA signal", MPEG Meeting; Oct. 2014; Strasbourg; (Motion Picture Expert Group or ISO/IEC JTC1/SC29/WG11), No. m35160, XP030063532, 5 pp.

"WD1-HOA Text of MPEG-H 3D Audio", 107. MPEG Meeting; Jan. 2014; San Jose; (Motion Picture Expert Group or ISO/IEC JTC1/SC29/WG11), No. N14264, XP030021001, 84 pp.

"Call for Proposals for 3D Audio," ISO/IEC JTC1/SC29/WG11/N13411, Geneva, CH; Jan. 2013, 20 pp.

Herre, et al., "MPEG-H 3D Audio—The New Standard for Coding of Immersive Spatial Audio," IEEE Journal of Selected Topics in Signal Processing, vol. 9, No. 5, Aug. 2015, pp. 770-779.

Poletti, "Three-Dimensional Surround Sound Systems Based on Spherical Harmonics," J. Audio Eng. Soc., vol. 53, No. 11, Nov. 2005, pp. 1004-1025.

"Information technology—High efficiency coding and media delivery in heterogeneous environments—Part 3: Part 3: 3D Audio, Amendment 3: MPEG-H 3D Audio Phase 2," ISO/IEC JTC 1/SC 29/WG 11, Jul. 25, 2015, 208 pp.

"Information technology—High efficiency coding and media delivery in heterogeneous environments—Part 3: 3D Audio," ISO/IEC JTC 1/SC 29/ WG 11, Apr. 4, 2014, 337 pp.

"Information technology—High efficiency coding and media delivery in heterogeneous environments—Part 3: 3D Audio," ISO/IEC JTC 1/SC 29, Jul. 25, 2014, 311 pp.

International Search Report and Written Opinion from International Application No. PCT/US2015/031195, dated Oct. 26, 2015, 18 pp.

Second Written Opinion from International Application No. PCT/US2015/031195, dated May 20, 2016, 7 pp.

Response to Second Written Opinion dated May 20, 2016, from International Application No. PCT/US2015/031195, filed on Jul. 20, 2016, 5 pp.

Information technology—MPEG audio technologies—Part 3: Unified speech and audio coding, First edition, Apr. 1, 2012, 286 pp.

International Preliminary Report on Patentability from International Application No. PCT/US2015/031195, dated Sep. 8, 2016, 8 pp.

* cited by examiner

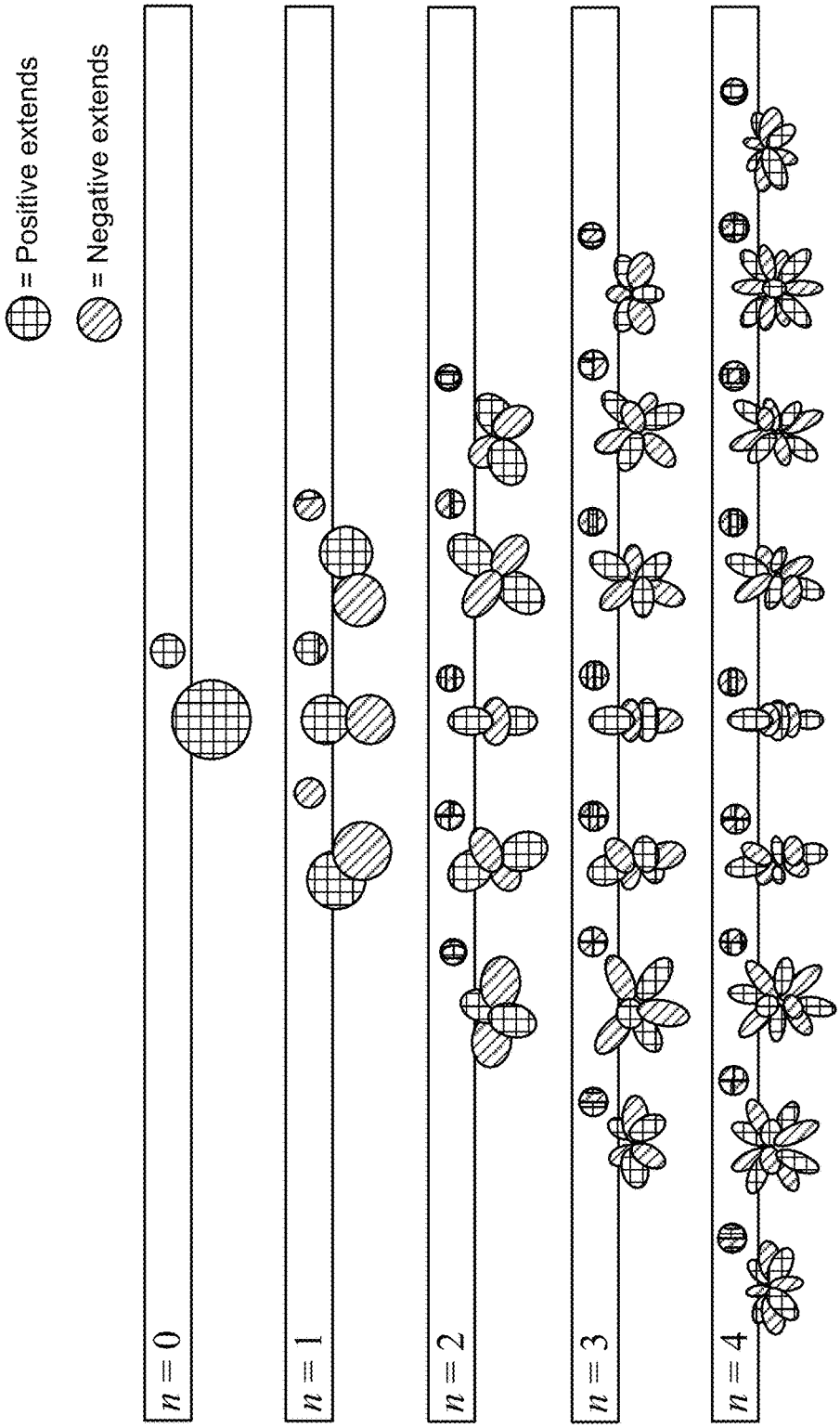


FIG. 1

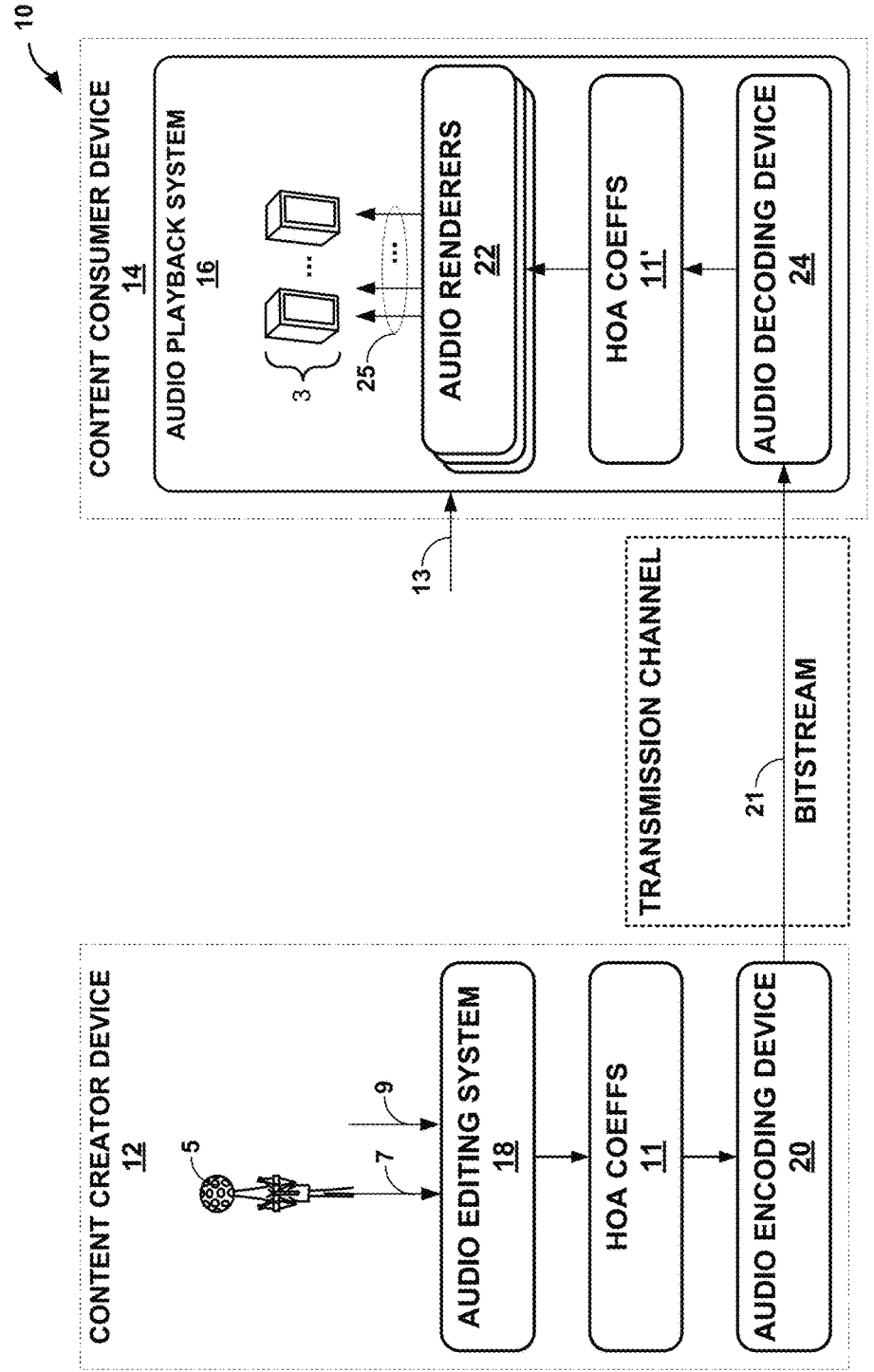

10
CONTENT CREATOR DEVICE
12
5
7
9
AUDIO EDITING SYSTEM
18
HOA COEFFS
11
AUDIO ENCODING DEVICE
20
TRANSMISSION CHANNEL
21
BITSTREAM
13
CONTENT CONSUMER DEVICE
14
AUDIO PLAYBACK SYSTEM
16
3
25
AUDIO RENDERERS
22
HOA COEFFS
11'
AUDIO DECODING DEVICE
24

FIG. 2

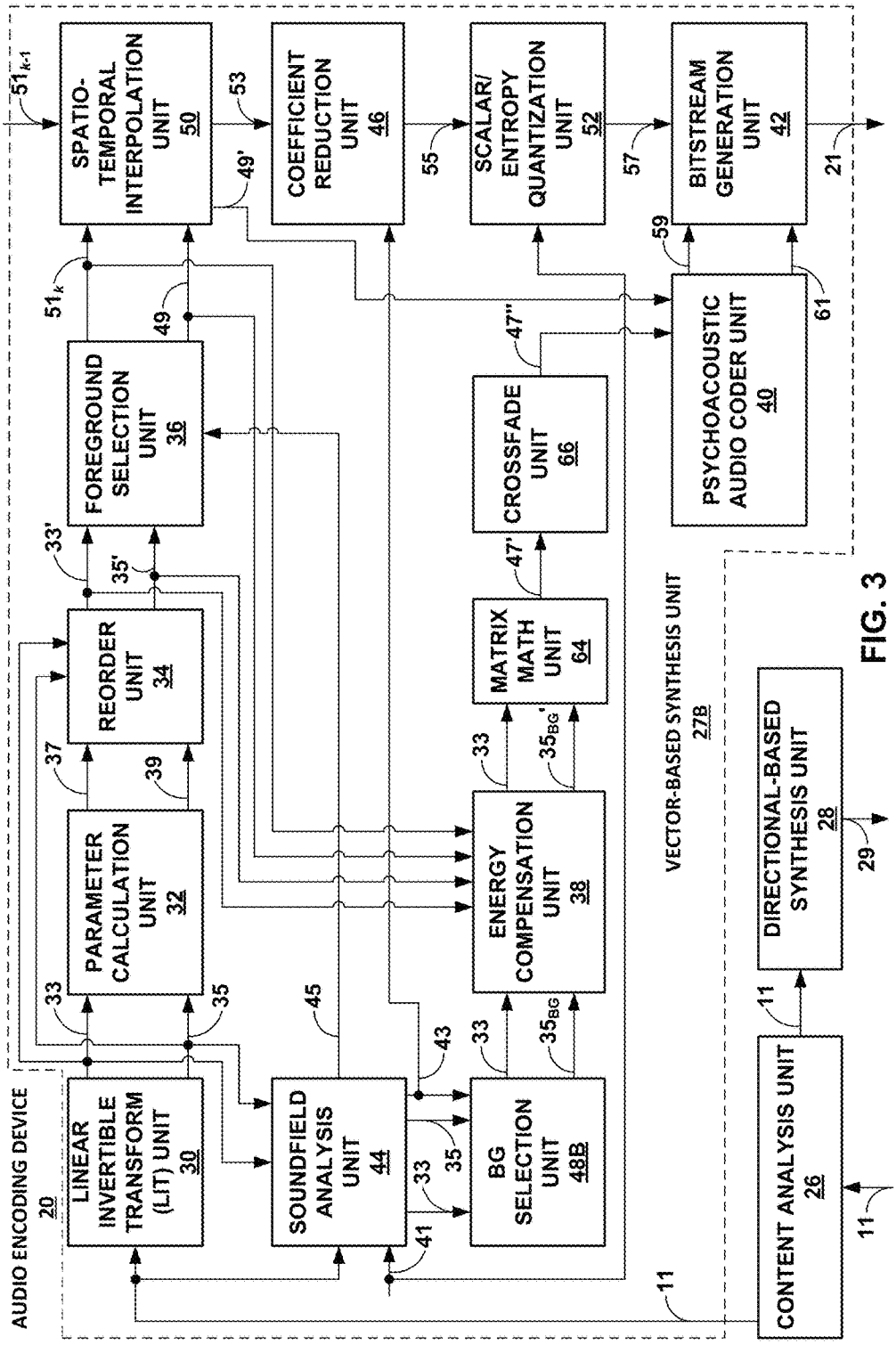

AUDIO ENCODING DEVICE
20
VECTOR-BASED SYNTHESIS UNIT
27B
LINEAR INVERTIBLE TRANSFORM (LIT) UNIT
30
PARAMETER CALCULATION UNIT
32
REORDER UNIT
34
SOUNDFIELD ANALYSIS UNIT
44
BG SELECTION UNIT
48B
ENERGY COMPENSATION UNIT
38
MATRIX MATH UNIT
64
CROSSFADE UNIT
66
FOREGROUND SELECTION UNIT
36
SPATIO-TEMPORAL INTERPOLATION UNIT
50
COEFFICIENT REDUCTION UNIT
46
SCALAR/ ENTROPY QUANTIZATION UNIT
52
BITSTREAM GENERATION UNIT
42
PSYCHOACOUSTIC AUDIO CODER UNIT
40
CONTENT ANALYSIS UNIT
26
DIRECTIONAL-BASED SYNTHESIS UNIT
28
51_{k-1}
51_k
53
49'
49
55
57
59
61
21
33'
35'
37
39
33
35
45
43
41
35_{BG}
$35_{BG}'$
47'
47''
11
29

FIG. 3

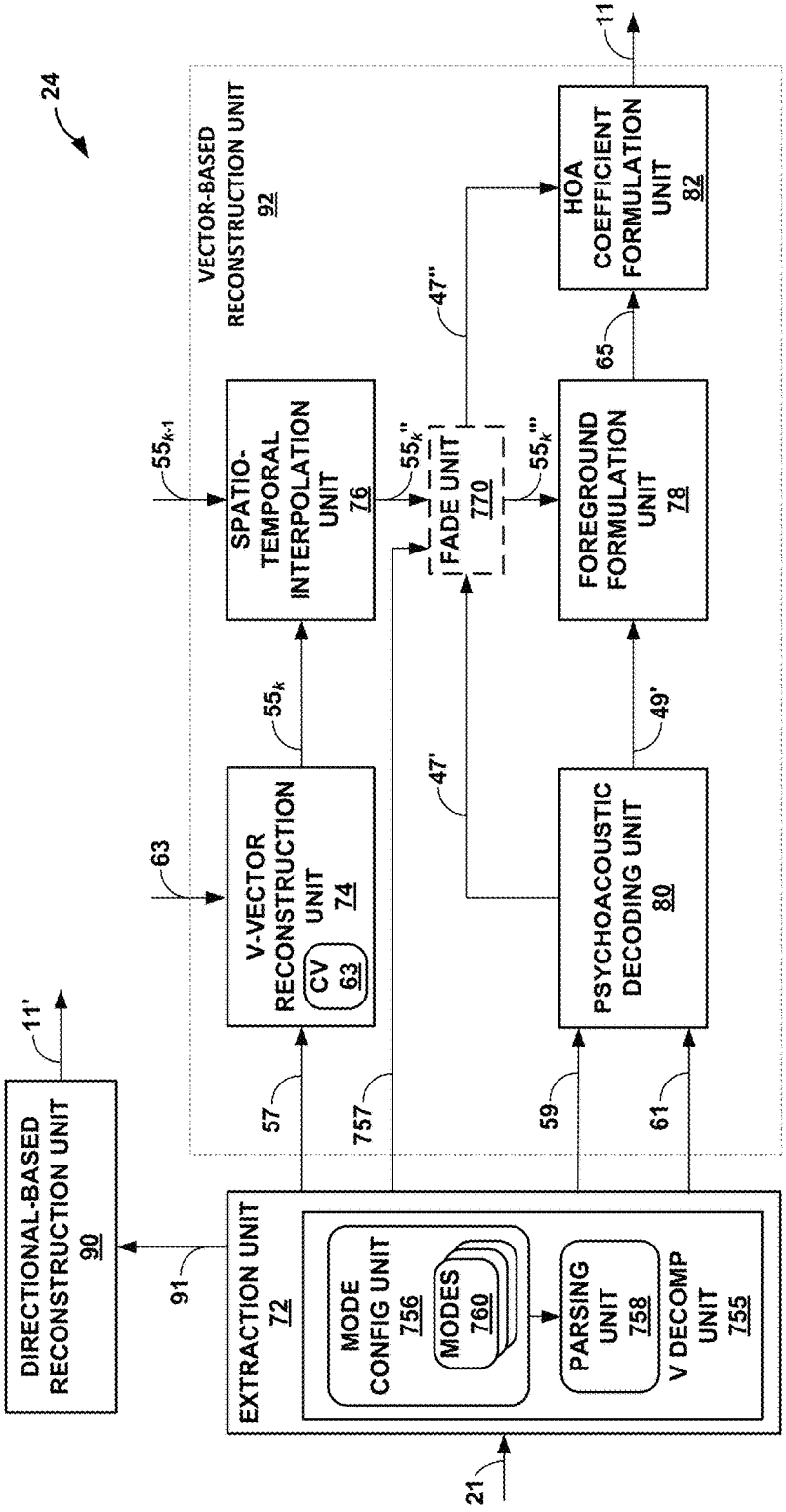

24
VECTOR-BASED RECONSTRUCTION UNIT 92
11'
HOA COEFFICIENT FORMULATION UNIT 82
65
47''
55_{k-1}
SPATIO-TEMPORAL INTERPOLATION UNIT 76
$55_k''$
FADE UNIT 770
$55_k'''$
FOREGROUND FORMULATION UNIT 78
55_k
63
V-VECTOR RECONSTRUCTION UNIT 74
CV 63
47'
49'
PSYCHOACOUSTIC DECODING UNIT 80
DIRECTIONAL-BASED RECONSTRUCTION UNIT 90
11'
57
757
59
61
91
EXTRACTION UNIT 72
MODE CONFIG UNIT 756
MODES 760
PARSING UNIT 758
V DECOMP UNIT 755
21

FIG. 4

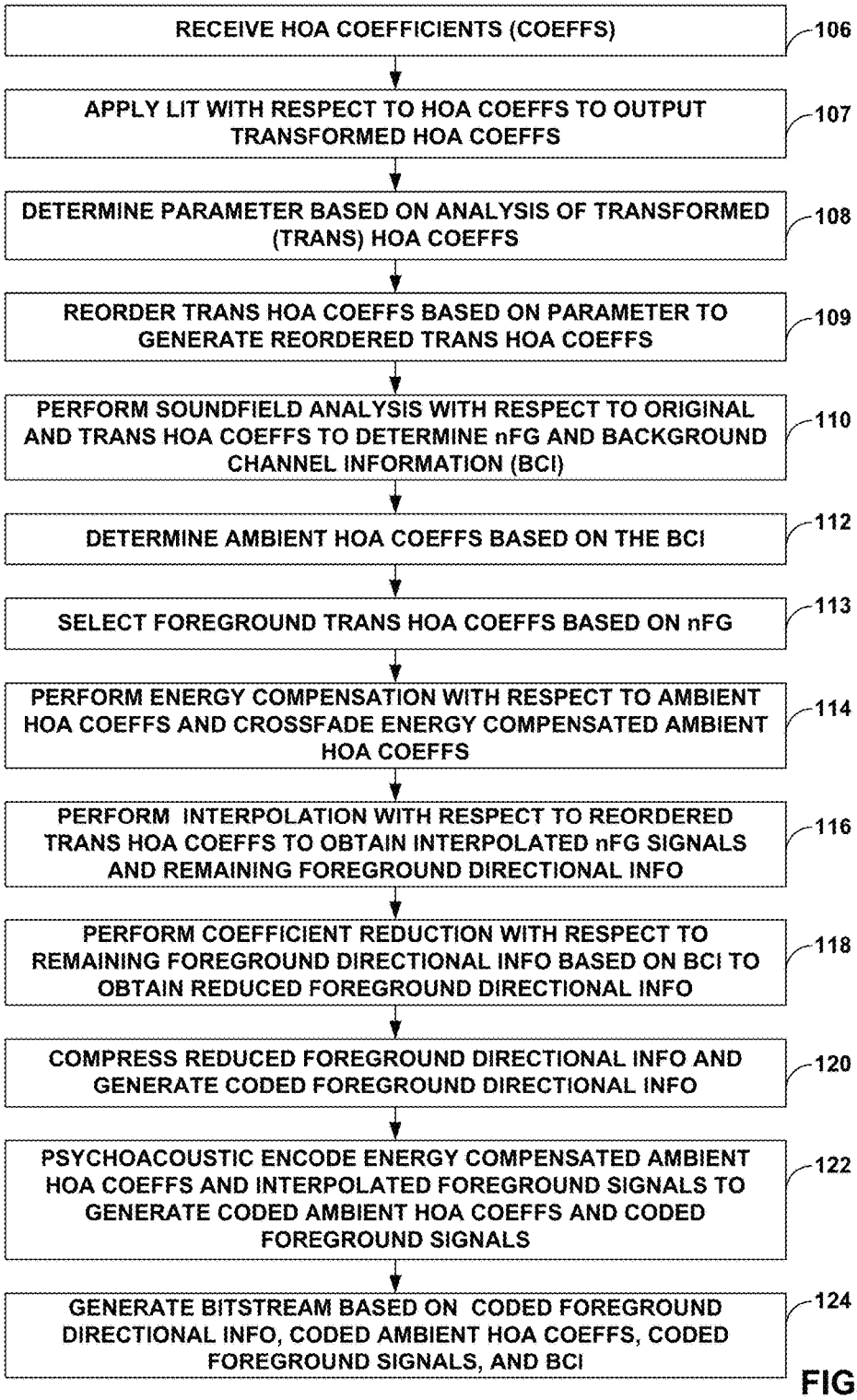

RECEIVE HOA COEFFICIENTS (COEFFS)
106
APPLY LIT WITH RESPECT TO HOA COEFFS TO OUTPUT TRANSFORMED HOA COEFFS
107
DETERMINE PARAMETER BASED ON ANALYSIS OF TRANSFORMED (TRANS) HOA COEFFS
108
REORDER TRANS HOA COEFFS BASED ON PARAMETER TO GENERATE REORDERED TRANS HOA COEFFS
109
PERFORM SOUNDFIELD ANALYSIS WITH RESPECT TO ORIGINAL AND TRANS HOA COEFFS TO DETERMINE nFG AND BACKGROUND CHANNEL INFORMATION (BCI)
110
DETERMINE AMBIENT HOA COEFFS BASED ON THE BCI
112
SELECT FOREGROUND TRANS HOA COEFFS BASED ON nFG
113
PERFORM ENERGY COMPENSATION WITH RESPECT TO AMBIENT HOA COEFFS AND CROSSFADE ENERGY COMPENSATED AMBIENT HOA COEFFS
114
PERFORM INTERPOLATION WITH RESPECT TO REORDERED TRANS HOA COEFFS TO OBTAIN INTERPOLATED nFG SIGNALS AND REMAINING FOREGROUND DIRECTIONAL INFO
116
PERFORM COEFFICIENT REDUCTION WITH RESPECT TO REMAINING FOREGROUND DIRECTIONAL INFO BASED ON BCI TO OBTAIN REDUCED FOREGROUND DIRECTIONAL INFO
118
COMPRESS REDUCED FOREGROUND DIRECTIONAL INFO AND GENERATE CODED FOREGROUND DIRECTIONAL INFO
120
PSYCHOACOUSTIC ENCODE ENERGY COMPENSATED AMBIENT HOA COEFFS AND INTERPOLATED FOREGROUND SIGNALS TO GENERATE CODED AMBIENT HOA COEFFS AND CODED FOREGROUND SIGNALS
122
GENERATE BITSTREAM BASED ON CODED FOREGROUND DIRECTIONAL INFO, CODED AMBIENT HOA COEFFS, CODED FOREGROUND SIGNALS, AND BCI
124

FIG. 5

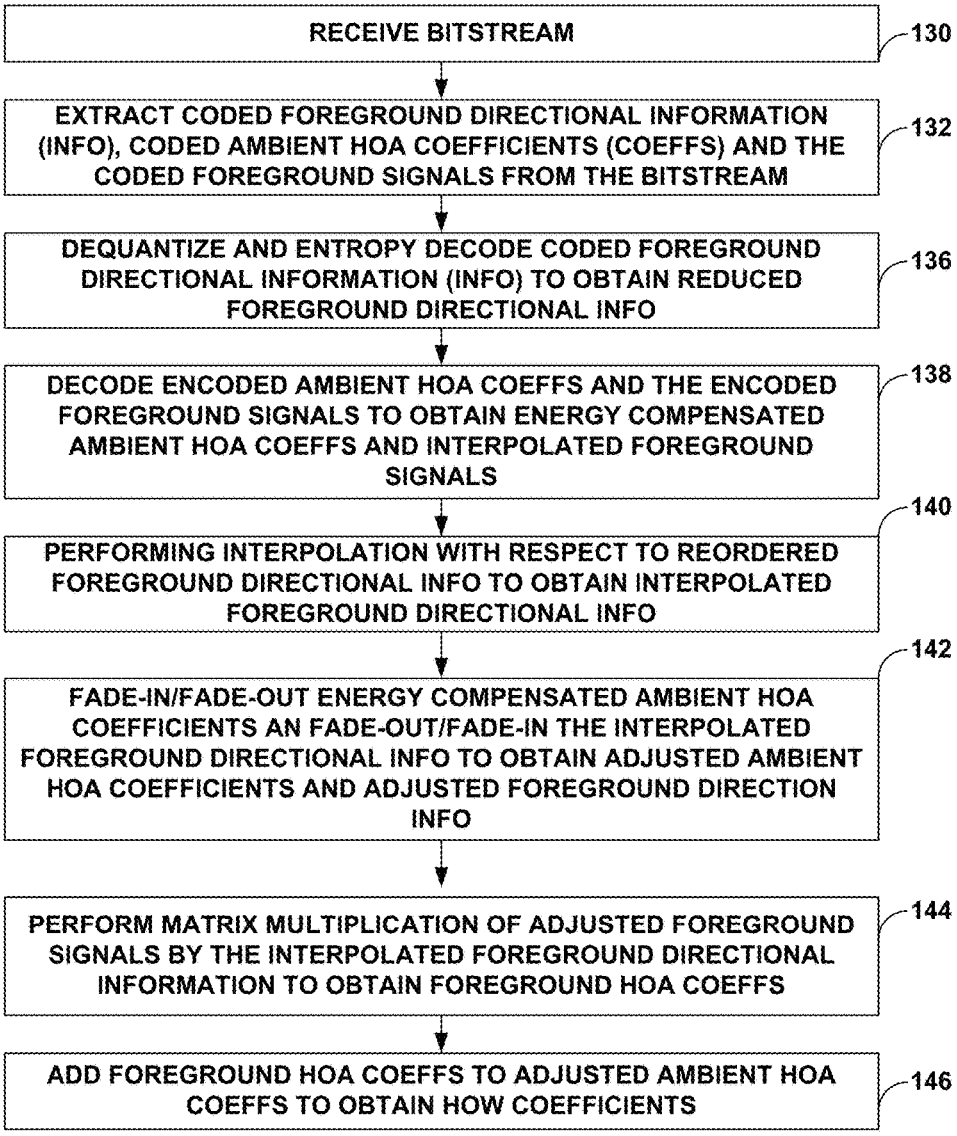

RECEIVE BITSTREAM
130
EXTRACT CODED FOREGROUND DIRECTIONAL INFORMATION (INFO), CODED AMBIENT HOA COEFFICIENTS (COEFFS) AND THE CODED FOREGROUND SIGNALS FROM THE BITSTREAM
132
DEQUANTIZE AND ENTROPY DECODE CODED FOREGROUND DIRECTIONAL INFORMATION (INFO) TO OBTAIN REDUCED FOREGROUND DIRECTIONAL INFO
136
DECODE ENCODED AMBIENT HOA COEFFS AND THE ENCODED FOREGROUND SIGNALS TO OBTAIN ENERGY COMPENSATED AMBIENT HOA COEFFS AND INTERPOLATED FOREGROUND SIGNALS
138
PERFORMING INTERPOLATION WITH RESPECT TO REORDERED FOREGROUND DIRECTIONAL INFO TO OBTAIN INTERPOLATED FOREGROUND DIRECTIONAL INFO
140
FADE-IN/FADE-OUT ENERGY COMPENSATED AMBIENT HOA COEFFICIENTS AN FADE-OUT/FADE-IN THE INTERPOLATED FOREGROUND DIRECTIONAL INFO TO OBTAIN ADJUSTED AMBIENT HOA COEFFICIENTS AND ADJUSTED FOREGROUND DIRECTION INFO
142
PERFORM MATRIX MULTIPLICATION OF ADJUSTED FOREGROUND SIGNALS BY THE INTERPOLATED FOREGROUND DIRECTIONAL INFORMATION TO OBTAIN FOREGROUND HOA COEFFS
144
ADD FOREGROUND HOA COEFFS TO ADJUSTED AMBIENT HOA COEFFS TO OBTAIN HOW COEFFICIENTS
146

FIG. 6

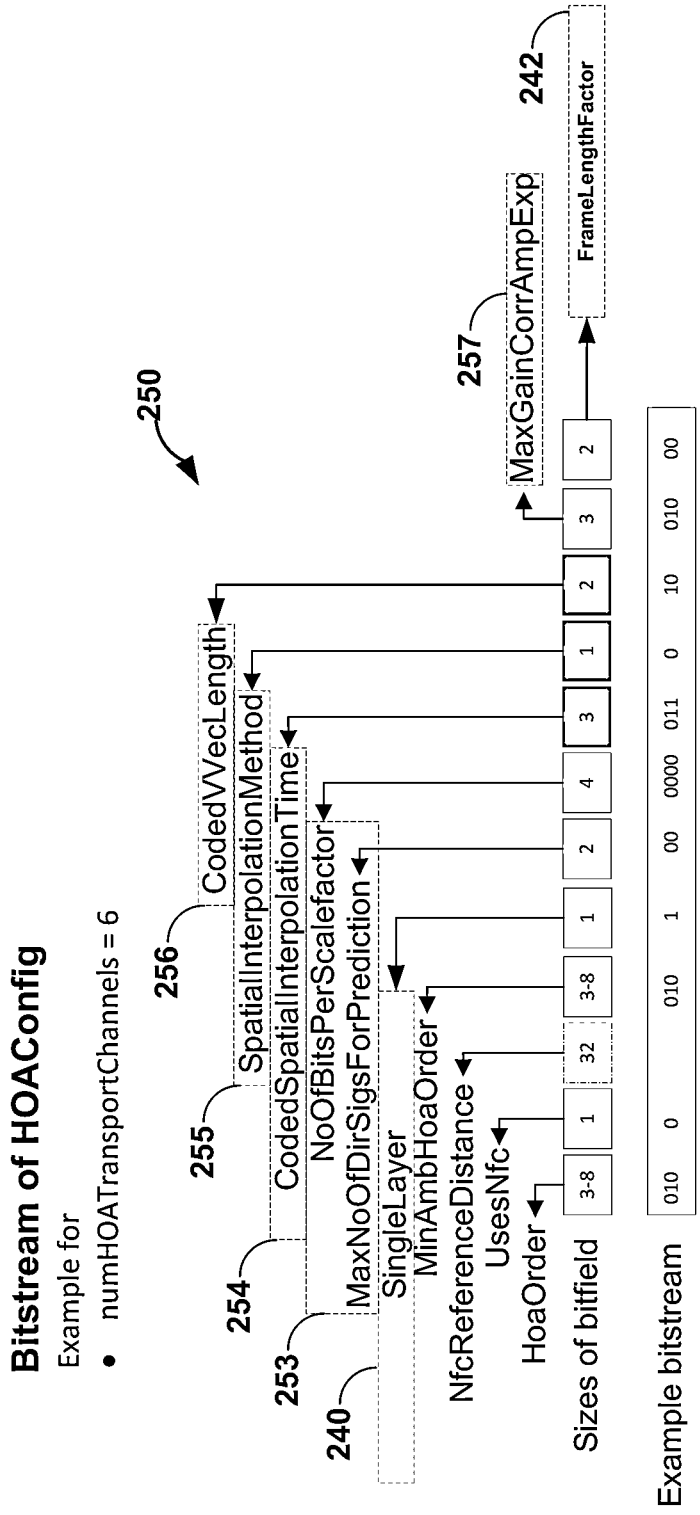

Bitstream of HOAConfig
Example for
• numHOATransportChannels = 6
250
256
CodedVVecLength
255
SpatialInterpolationMethod
254
CodedSpatialInterpolationTime
253
NoOfBitsPerScalefactor
240
MaxNoOfDirSigsForPrediction
SingleLayer
MinAmbHoaOrder
NfcReferenceDistance
UsesNfc
HoaOrder
Sizes of bitfield
Example bitstream
3-8
1
32
3-8
1
2
4
3
1
2
3
2
257
MaxGainCorrAmpExp
242
FrameLengthFactor
010
0
010
1
00
0000
011
0
10
010
00

FIG. 7

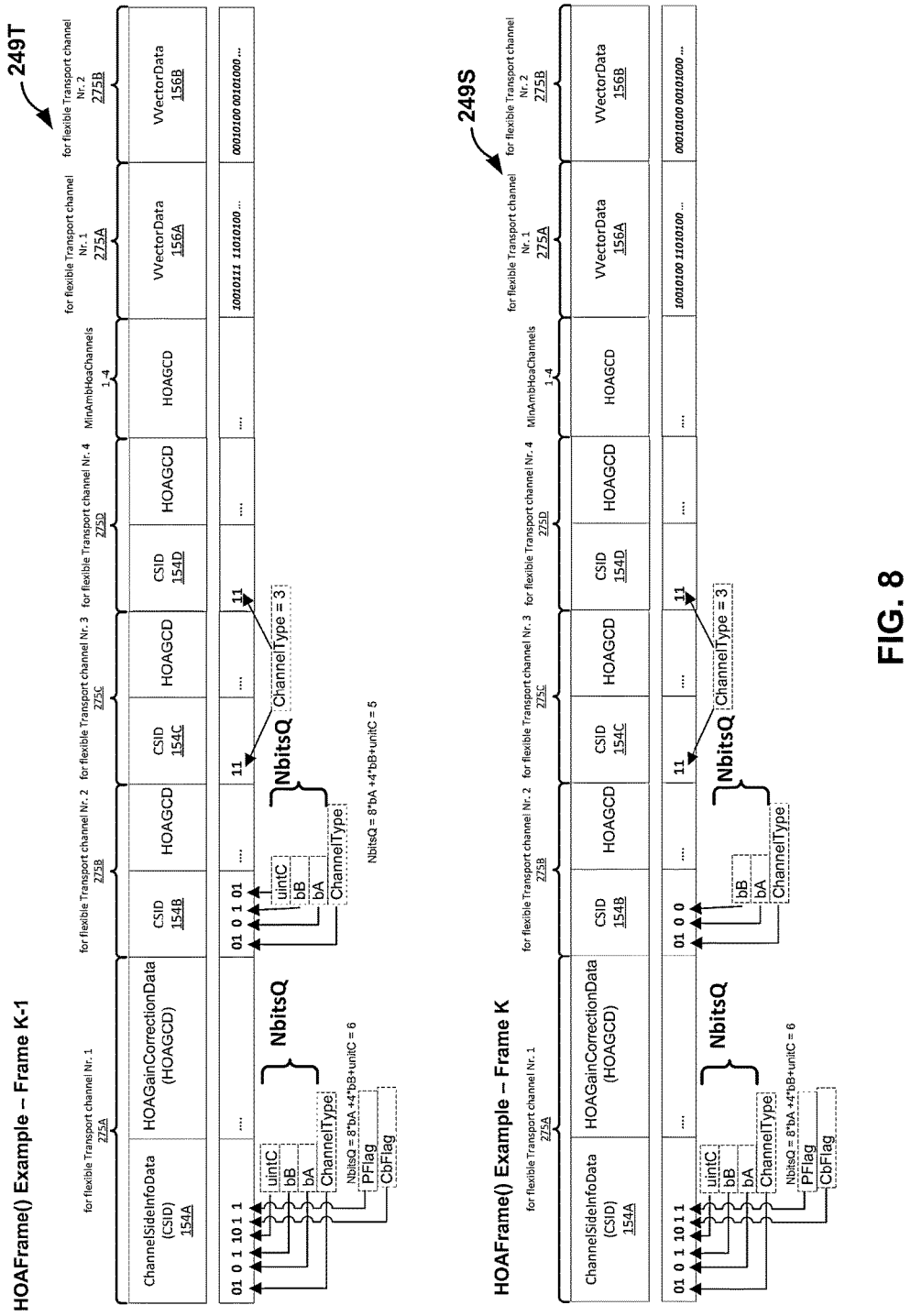

249T
HOAFrame() Example – Frame K-1
for flexible Transport channel Nr. 1 275A
for flexible Transport channel Nr. 2 275B
for flexible Transport channel Nr. 3 275C
for flexible Transport channel Nr. 4 275D
MinAmbHoaChannels 1..4
for flexible Transport channel Nr. 1 275A
for flexible Transport channel Nr. 2 275B
ChannelSideInfoData (CSID) 154A
HOAGainCorrectionData (HOAGCD)
CSID 154B
HOAGCD
CSID 154C
HOAGCD
CSID 154D
HOAGCD
HOAGCD
VVectorData 156A
VVectorData 156B
01 0 1 10 1 1
....
01 0 1 01
....
11
....
11
....
....
10010111 11010100 ...
00010100 00101000 ...
NbitsQ
uintC
bB
bA
ChannelType
ChannelType = 3
NbitsQ = 8*bA +4*bB+uintC = 6
PFlag
CbFlag
NbitsQ = 8*bA +4*bB+uintC = 5
249S
HOAFrame() Example – Frame K
for flexible Transport channel Nr. 1 275A
for flexible Transport channel Nr. 2 275B
for flexible Transport channel Nr. 3 275C
for flexible Transport channel Nr. 4 275D
MinAmbHoaChannels 1..4
for flexible Transport channel Nr. 1 275A
for flexible Transport channel Nr. 2 275B
ChannelSideInfoData (CSID) 154A
HOAGainCorrectionData (HOAGCD)
CSID 154B
HOAGCD
CSID 154C
HOAGCD
CSID 154D
HOAGCD
HOAGCD
VVectorData 156A
VVectorData 156B
01 0 1 10 1 1
....
01 0 0
....
11
....
11
....
....
10010100 11010100 ...
00010100 00101000 ...
NbitsQ
uintC
bB
bA
ChannelType
ChannelType = 3
NbitsQ = 8*bA +4*bB+uintC = 6
PFlag
CbFlag

FIG. 8

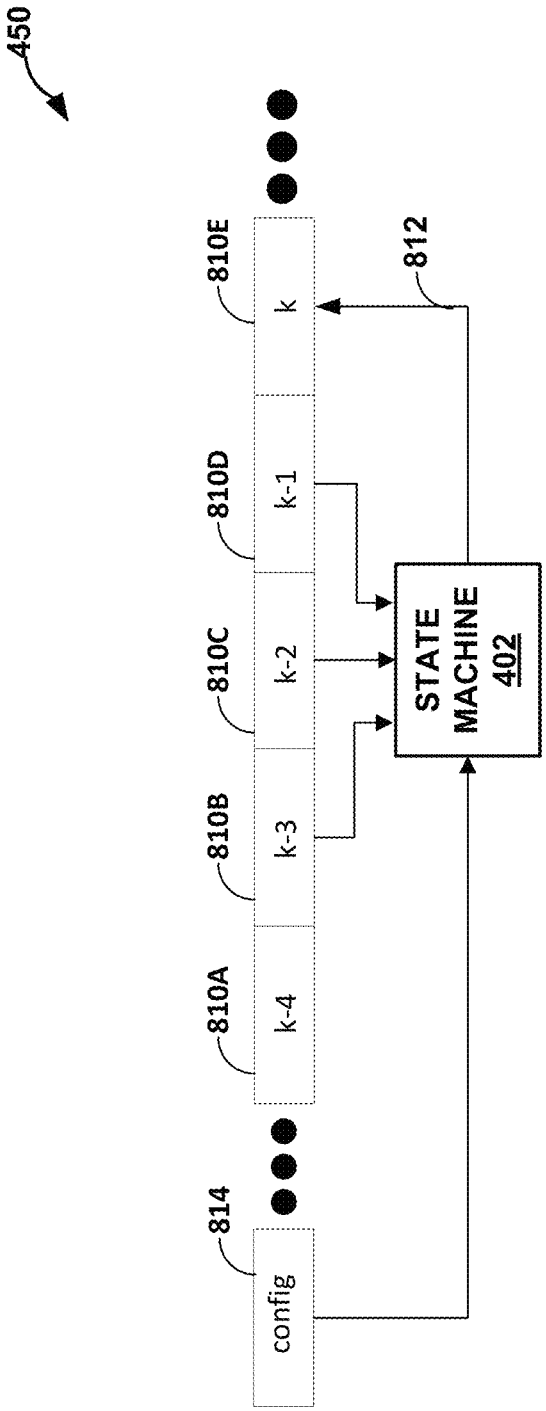

450
810E
k
812
810D
k-1
810C
k-2
STATE MACHINE 402
810B
k-3
810A
k-4
814
config

FIG. 9

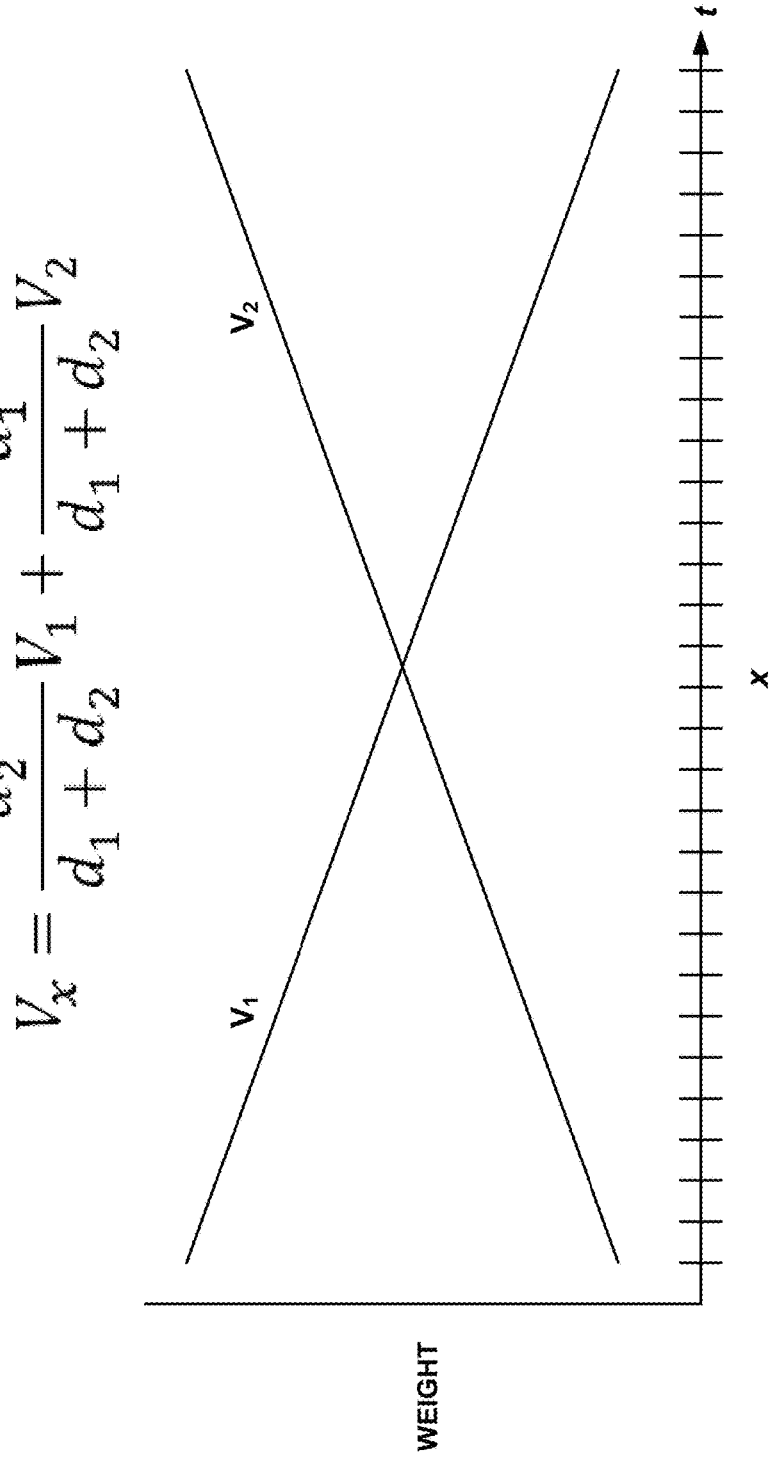


FIG. 10

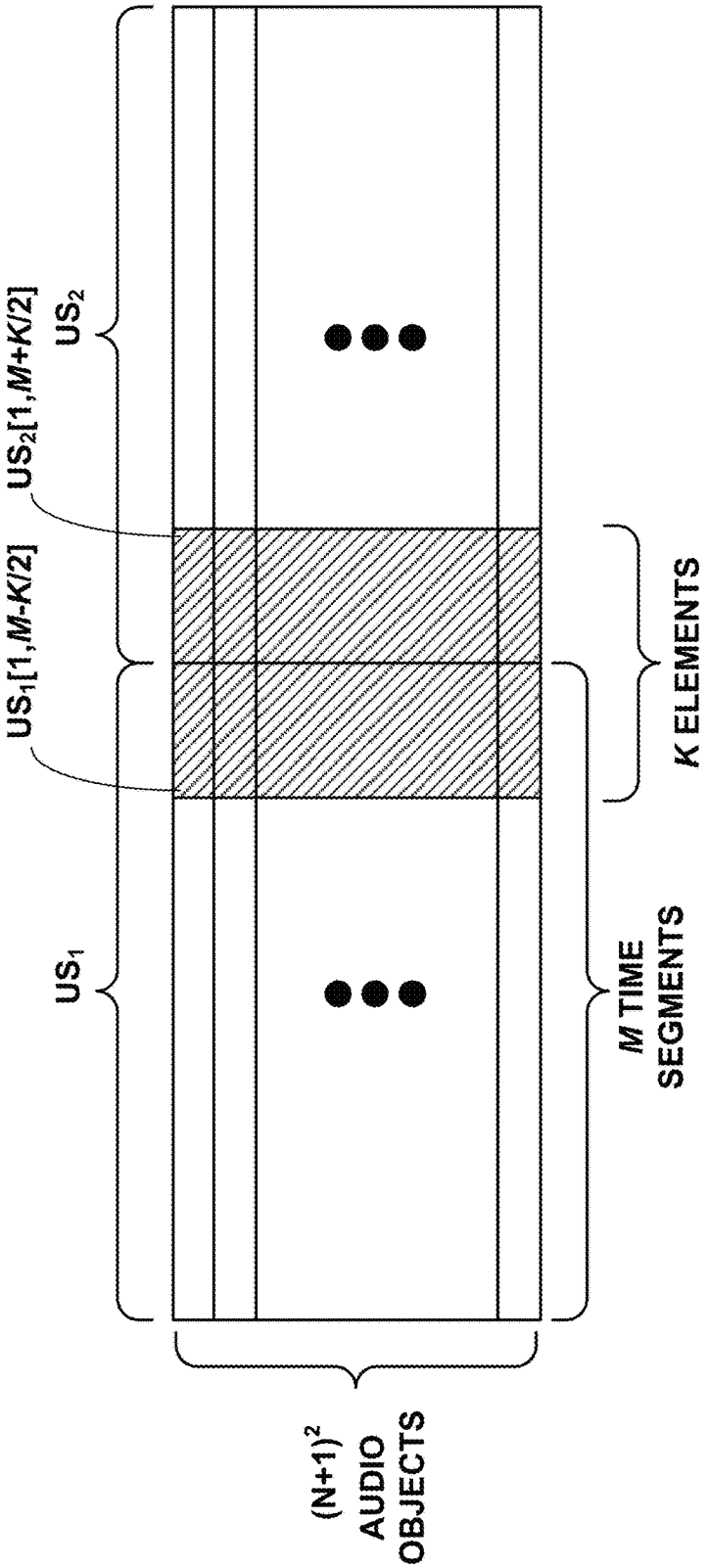

US2
US2[1,M+K/2]
US1[1,M-K/2]
US1
K ELEMENTS
M TIME SEGMENTS
(N+1)2 AUDIO OBJECTS

FIG. 11

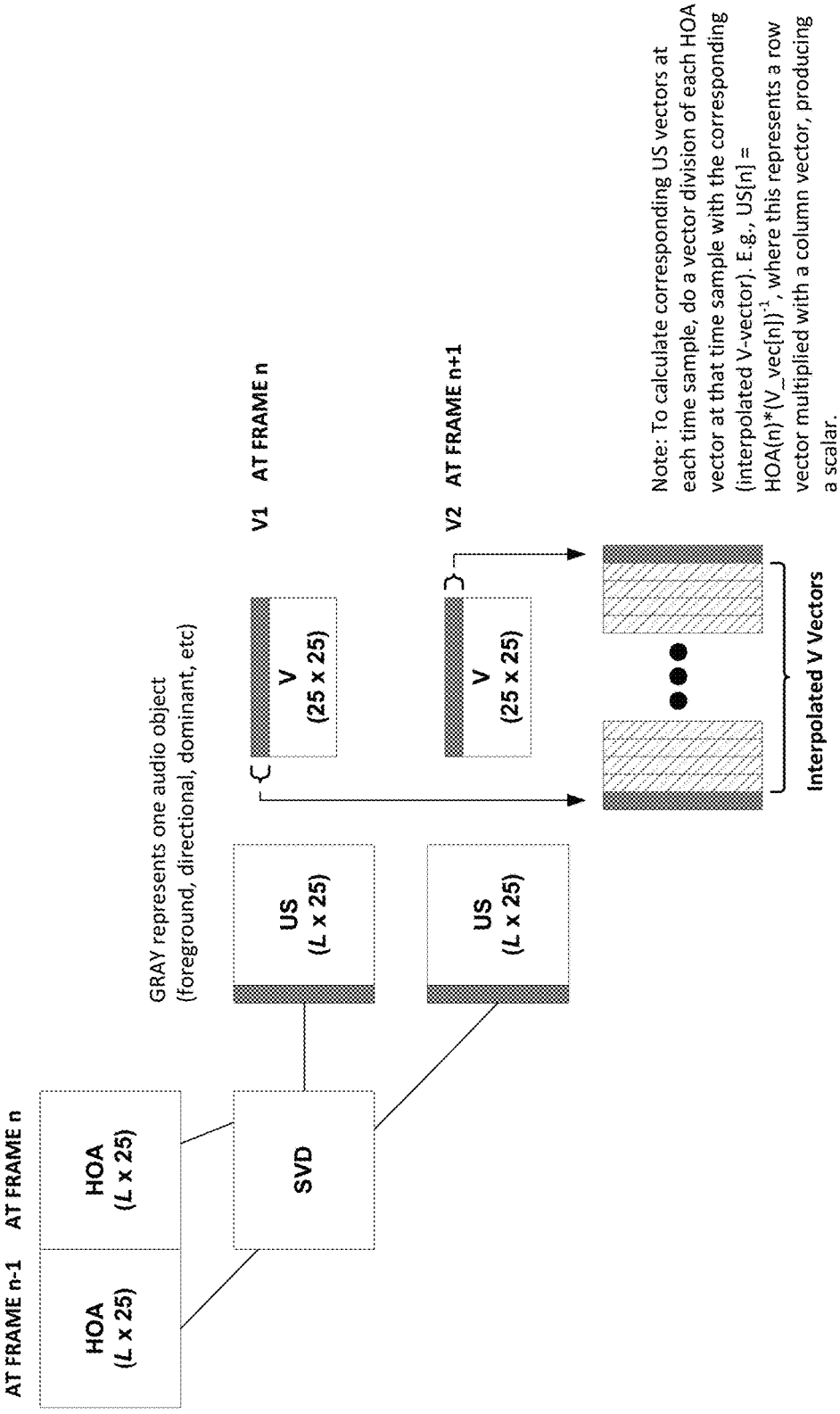

AT FRAME n-1
HOA (L x 25)
AT FRAME n
HOA (L x 25)
SVD
US (L x 25)
US (L x 25)
V (25 x 25)
V (25 x 25)
V1 AT FRAME n
V2 AT FRAME n+1
GRAY represents one audio object (foreground, directional, dominant, etc)
Interpolated V Vectors
Note: To calculate corresponding US vectors at each time sample, do a vector division of each HOA vector at that time sample with the corresponding (interpolated V-vector). E.g., US[n] = HOA(n)*(V_vec[n])$^{-1}$, where this represents a row vector multiplied with a column vector, producing a scalar.

FIG. 12

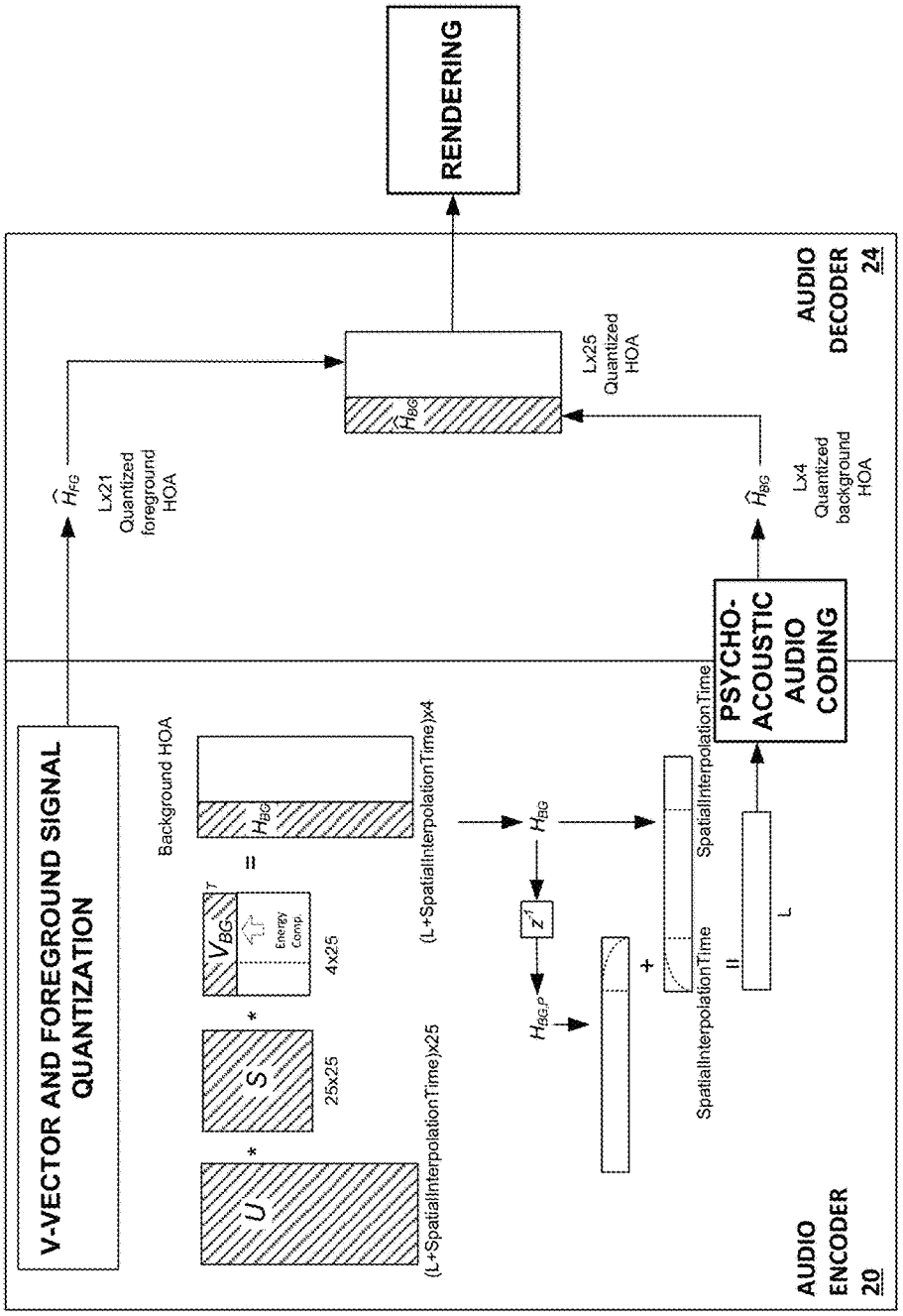

RENDERING
AUDIO DECODER 24
Lx25 Quantized HOA
Lx21 Quantized foreground HOA
Lx4 Quantized background HOA
PSYCHO-ACOUSTIC AUDIO CODING
V-VECTOR AND FOREGROUND SIGNAL QUANTIZATION
Background HOA
(L+SpatialInterpolationTime)x4
(L+SpatialInterpolationTime)x25
25x25
4x25
Energy Comp.
SpatialInterpolationTime
L
AUDIO ENCODER 20

FIG. 13

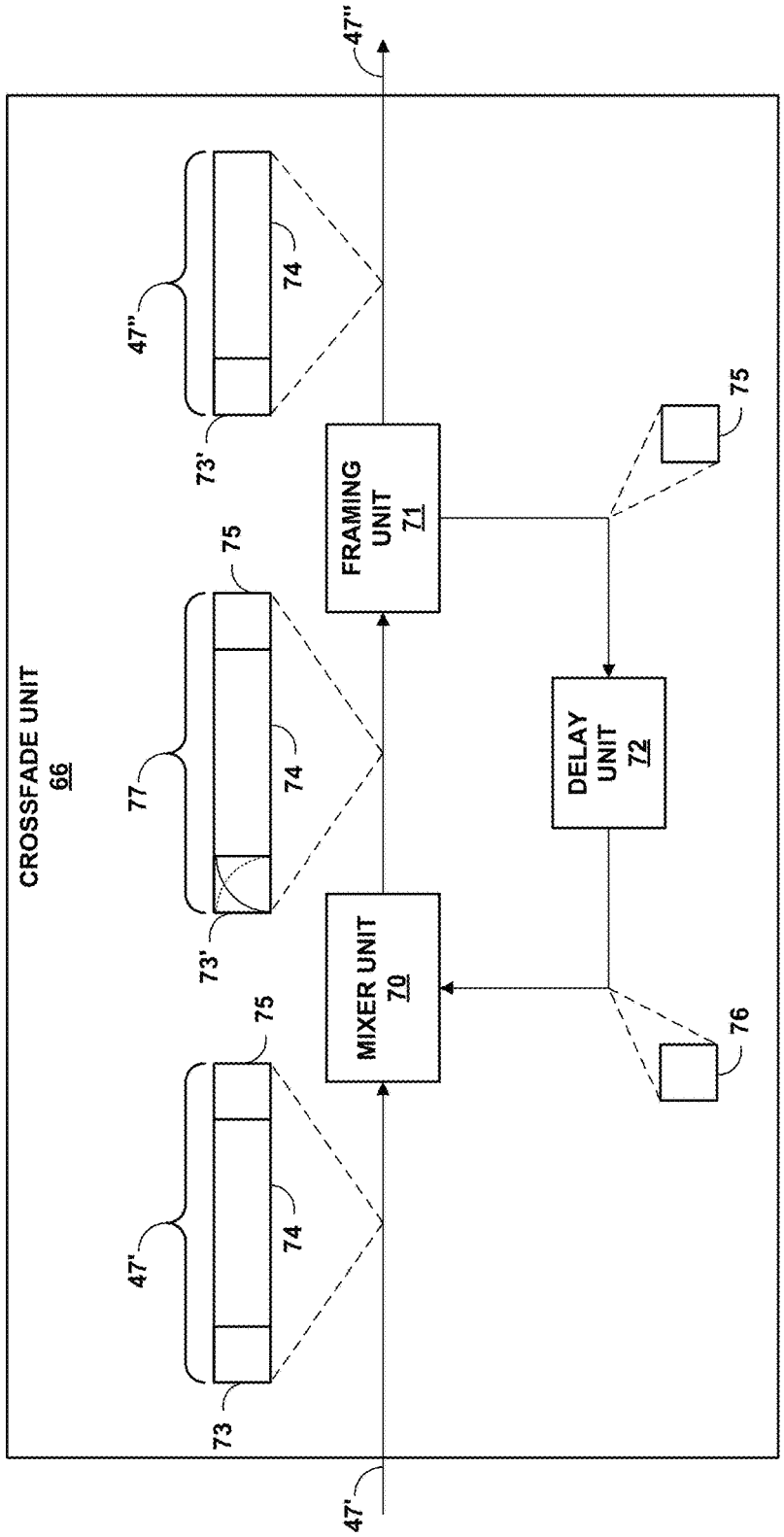

47''
CROSSFADE UNIT
66
47''
74
73'
75
FRAMING UNIT
71
75
77
74
73'
DELAY UNIT
72
MIXER UNIT
70
75
76
47'
74
73
47'

FIG. 14

CROSSFADING BETWEEN HIGHER ORDER AMBISONIC SIGNALS

This application claims the benefit of the following U.S. Provisional applications:

U.S. Provisional Application No. 61/994,763, filed May 16, 2014, entitled "CROSSFADING BETWEEN HIGHER ORDER AMBISONIC SIGNALS;"

U.S. Provisional Application No. 62/004,076, filed May 28, 2014, entitled "CROSSFADING BETWEEN HIGHER ORDER AMBISONIC SIGNALS;" and

U.S. Provisional Application No. 62/118,434, filed Feb. 19, 2015, entitled "CROSSFADING BETWEEN HIGHER ORDER AMBISONIC SIGNALS,"

where each of the foregoing listed U.S. Provisional applications is incorporated by reference as if set forth in their respective entirety herein.

TECHNICAL FIELD

This disclosure relates to audio data and, more specifically, coding of higher-order ambisonic audio data.

BACKGROUND

A higher order ambisonics (HOA) signal (often represented by a plurality of spherical harmonic coefficients (SHC) or other hierarchical elements) is a three-dimensional representation of a soundfield. This HOA or SHC representation may represent this soundfield in a manner that is independent of the local speaker geometry used to playback a multi-channel audio signal rendered from this SHC signal. This SHC signal may also facilitate backwards compatibility as this SHC signal may be rendered to well-known and highly adopted multi-channel formats, such as a 5.1 audio channel format or a 7.1 audio channel format. The SHC representation may therefore enable a better representation of a soundfield that also accommodates backward compatibility.

SUMMARY

In general, techniques are described for crossfading between ambient HOA coefficients. For example, techniques are described for crossfading between a current set of ambient HOA coefficients and a previous set of ambient HOA coefficients in the energy compensated domain. In this way, the techniques of this disclosure may smooth the transition between the previous set of ambient HOA coefficients and the current set of ambient HOA coefficients.

In one aspect, a method includes crossfading, by a device, between a first set of ambient spherical harmonic coefficients (SHCs) and a second set of ambient SHCs to obtain a first set of crossfaded ambient SHCs, wherein the first set of SHCs describe a first sound field and the second set of SHCs describe a second sound field.

In another aspect, a device includes one or more processors; and at least one module executable by the one or more processors to crossfade between a first set of ambient SHCs and a second set of ambient SHCs to obtain a first set of crossfaded ambient SHCs, wherein the first set of SHCs describe a first sound field and the second set of SHCs describe a second sound field.

In another aspect, a device includes means for obtaining a first set of ambient SHCs, wherein the first set of SHCs describe a first sound field; means for obtaining a second set of ambient SHCs, wherein the second set of SHCs describe a second sound field; and means for crossfading between the first set of ambient SHCs and the second set of ambient SHCs to obtain a first set of crossfaded ambient SHCs.

In another aspect, a computer-readable storage medium stores instructions that, when executed, cause one or more processors of a device to crossfade between a first set of ambient SHCs and a second set of ambient SHCs to obtain a first set of crossfaded ambient SHCs, wherein the first set of SHCs describe a first sound field and the second set of SHCs describe a second sound field.

In another aspect, a method comprises crossfading, by a device, between a first set of spherical harmonic coefficients (SHCs) and a second set of SHCs to obtain a first set of crossfaded SHCs, wherein the first set of SHCs describe a first sound field and the second set of SHCs describe a second sound field.

In another aspect, an audio decoding device comprises a memory configured to store a first set of spherical harmonic coefficients (SHCs) and a second set of SHCs, wherein the first set of SHCs describe a first sound field and the second set of SHCs describe a second sound field. The audio decoding device further comprises one or more processors configured to crossfade between the first set of SHCs and a second set of SHCs to obtain a first set of crossfaded ambient SHCs.

In another aspect, an audio encoding device comprises a memory configured to store a first set of spherical harmonic coefficients (SHCs) and a second set of SHCs, wherein the first set of SHCs describe a first sound field and the second set of SHCs describe a second sound field. The audio encoding device also comprises one or more processors configured to crossfade between the first set of SHCs and a second set of SHCs to obtain a first set of crossfaded SHCs.

In another aspect, an apparatus comprises means for storing a first set of spherical harmonic coefficients (SHCs) and a second set of SHCs, wherein the first set of SHCs describe a first sound field and the second set of SHCs describe a second sound field, and means for crossfading between the first set of SHCs and a second set of SHCs to obtain a first set of crossfaded SHCs.

The details of one or more aspects of the techniques are set forth in the accompanying drawings and the description below. Other features, objects, and advantages of these techniques will be apparent from the description and drawings, and from the claims.

BRIEF DESCRIPTION OF DRAWINGS

FIG. **1** is a diagram illustrating spherical harmonic basis functions of various orders and sub-orders.

FIG. **2** is a diagram illustrating a system that may perform various aspects of the techniques described in this disclosure.

FIG. **3** is a block diagram illustrating, in more detail, one example of the audio encoding device shown in the example of FIG. **2** that may perform various aspects of the techniques described in this disclosure.

FIG. **4** is a block diagram illustrating the audio decoding device of FIG. **2** in more detail.

FIG. **5** is a flowchart illustrating exemplary operation of an audio encoding device in performing various aspects of the vector-based synthesis techniques described in this disclosure.

FIG. **6** is a flowchart illustrating exemplary operation of an audio decoding device in performing various aspects of the techniques described in this disclosure.

FIGS. **7** and **8** are diagrams illustrating, in more detail, the bitstream that may specify the compressed spatial components.

FIG. **9** is a diagram illustrating a portion of the bitstream that may specify the compressed spatial components in more detail.

FIG. **10** illustrates a representation of techniques for obtaining a spatio-temporal interpolation as described herein.

FIG. **11** is a block diagram illustrating artificial US matrices, US_1 and US_2, for sequential SVD blocks for a multi-dimensional signal according to techniques described herein.

FIG. **12** is a block diagram illustrating decomposition of subsequent frames of a higher-order ambisonics (HOA) signal using Singular Value Decomposition and smoothing of the spatio-temporal components according to techniques described in this disclosure.

FIG. **13** is a diagram illustrating one or more an audio encoder and an audio decoder configured to perform one or more techniques described in this disclosure.

FIG. **14** is a block diagram illustrating, in more detail, the crossfade unit of the audio encoding device shown in the example of FIG. **3**.

DETAILED DESCRIPTION

The evolution of surround sound has made available many output formats for entertainment nowadays. Examples of such consumer surround sound formats are mostly 'channel' based in that they implicitly specify feeds to loudspeakers in certain geometrical coordinates. The consumer surround sound formats include the popular 5.1 format (which includes the following six channels: front left (FL), front right (FR), center or front center, back left or surround left, back right or surround right, and low frequency effects (LFE)), the growing 7.1 format, various formats that includes height speakers such as the 7.1.4 format and the 22.2 format (e.g., for use with the Ultra High Definition Television standard). Non-consumer formats can span any number of speakers (in symmetric and non-symmetric geometries) often termed 'surround arrays'. One example of such an array includes 32 loudspeakers positioned on coordinates on the corners of a truncated icosahedron.

The input to a future MPEG encoder is optionally one of three possible formats: (i) traditional channel-based audio (as discussed above), which is meant to be played through loudspeakers at pre-specified positions; (ii) object-based audio, which involves discrete pulse-code-modulation (PCM) data for single audio objects with associated metadata containing their location coordinates (amongst other information); and (iii) scene-based audio, which involves representing the soundfield using coefficients of spherical harmonic basis functions (also called "spherical harmonic coefficients" or SHC, "Higher-order Ambisonics" or HOA, and "HOA coefficients"). The future MPEG encoder may be described in more detail in a document entitled "Call for Proposals for 3D Audio," by the International Organization for Standardization/International Electrotechnical Commission (ISO)/(IEC) JTC1/SC29/WG11/N13411, released January 2013 in Geneva, Switzerland, and available at http://mpeg.chiariglione.org/sites/default/files/files/standards/parts/docs/w13411.zip.

There are various 'surround-sound' channel-based formats in the market. They range, for example, from the 5.1 home theatre system (which has been the most successful in terms of making inroads into living rooms beyond stereo) to the 22.2 system developed by NHK (Nippon Hoso Kyokai or Japan Broadcasting Corporation). Content creators (e.g., Hollywood studios) would like to produce the soundtrack for a movie once, and not spend effort to remix it for each speaker configuration. Recently, Standards Developing Organizations have been considering ways in which to provide an encoding into a standardized bitstream and a subsequent decoding that is adaptable and agnostic to the speaker geometry (and number) and acoustic conditions at the location of the playback (involving a renderer).

To provide such flexibility for content creators, a hierarchical set of elements may be used to represent a soundfield. The hierarchical set of elements may refer to a set of elements in which the elements are ordered such that a basic set of lower-ordered elements provides a full representation of the modeled soundfield. As the set is extended to include higher-order elements, the representation becomes more detailed, increasing resolution.

One example of a hierarchical set of elements is a set of spherical harmonic coefficients (SHC). The following expression demonstrates a description or representation of a soundfield using SHC:

$$p_i(t, r_r, \theta_r, \varphi_r) = \sum_{\omega=0}^{\infty} \left[4\pi \sum_{n=0}^{\infty} j_n(kr_r) \sum_{m=-n}^{n} A_n^m(k) Y_n^m(\theta_r, \varphi_r) \right] e^{j\omega t},$$

The expression shows that the pressure p_i at any point $\{r_r, \theta_r, \varphi_r\}$ of the soundfield, at time t, can be represented uniquely by the SHC, $A_n^m(k)$. Here, $k=\omega/c$, c is the speed of sound (~343 m/s), $\{r_r, \theta_r, \varphi_r\}$ is a point of reference (or observation point), $j_n(\bullet)$ is the spherical Bessel function of order n, and $Y_n^m(\theta_r, \varphi_r)$ are the spherical harmonic basis functions of order n and suborder m. It can be recognized that the term in square brackets is a frequency-domain representation of the signal (i.e., $S(\omega, r_r, \theta_r, \varphi_r)$) which can be approximated by various time-frequency transformations, such as the discrete Fourier transform (DFT), the discrete cosine transform (DCT), or a wavelet transform. Other examples of hierarchical sets include sets of wavelet transform coefficients and other sets of coefficients of multiresolution basis functions.

FIG. **1** is a diagram illustrating spherical harmonic basis functions from the zero order (n=0) to the fourth order (n=4). As can be seen, for each order, there is an expansion of suborders m which are shown but not explicitly noted in the example of FIG. **1** for ease of illustration purposes.

The SHC $A_n^m(k)$ can either be physically acquired (e.g., recorded) by various microphone array configurations or, alternatively, they can be derived from channel-based or object-based descriptions of the soundfield. The SHC represent scene-based audio, where the SHC may be input to an audio encoder to obtain encoded SHC that may promote more efficient transmission or storage. For example, a fourth-order representation involving $(1+4)^2$ (25, and hence fourth order) coefficients may be used.

As noted above, the SHC may be derived from a microphone recording using a microphone array. Various examples of how SHC may be derived from microphone arrays are described in Poletti, M., "Three-Dimensional Surround Sound Systems Based on Spherical Harmonics," J. Audio Eng. Soc., Vol. 53, No. 11, 2005 November, pp. 1004-1025.

To illustrate how the SHCs may be derived from an object-based description, consider the following equation.

The coefficients $A_n^m(k)$ for the soundfield corresponding to an individual audio object may be expressed as:

$$A_n^m(k)=g(\omega)(-4\pi ik)h_n^{(2)}(kr_s)Y_n^{m*}(\theta_s,\varphi_s),$$

where i is $\sqrt{-1}$, $h_n^2(\bullet)$ is the spherical Hankel function (of the second kind) of order n, and $\{r_s, \theta_s, \varphi_s\}$ is the location of the object. Knowing the object source energy $g(\omega)$ as a function of frequency (e.g., using time-frequency analysis techniques, such as performing a fast Fourier transform on the PCM stream) allows us to convert each PCM object and the corresponding location into the SHC $A_n^m(k)$. Further, it can be shown (since the above is a linear and orthogonal decomposition) that the $A_n^m(k)$ coefficients for each object are additive. In this manner, a multitude of PCM objects can be represented by the $A_n^m(k)$ coefficients (e.g., as a sum of the coefficient vectors for the individual objects). Essentially, the coefficients contain information about the soundfield (the pressure as a function of 3D coordinates), and the above represents the transformation from individual objects to a representation of the overall soundfield, in the vicinity of the observation point $\{r_r, \theta_r, \varphi_r\}$. The remaining figures are described below in the context of object-based and SHC-based audio coding.

FIG. **2** is a diagram illustrating a system **10** that may perform various aspects of the techniques described in this disclosure. As shown in the example of FIG. **2**, the system **10** includes a content creator device **12** and a content consumer device **14**. While described in the context of the content creator device **12** and the content consumer device **14**, the techniques may be implemented in any context in which SHCs (which may also be referred to as HOA coefficients) or any other hierarchical representation of a soundfield are encoded to form a bitstream representative of the audio data. Moreover, the content creator device **12** may represent any form of computing device capable of implementing the techniques described in this disclosure, including a handset (or cellular phone), a tablet computer, a smart phone, or a desktop computer to provide a few examples. Likewise, the content consumer device **14** may represent any form of computing device capable of implementing the techniques described in this disclosure, including a handset (or cellular phone), a tablet computer, a smart phone, a set-top box, or a desktop computer to provide a few examples.

The content creator device **12** may be operated by a movie studio or other entity that may generate multi-channel audio content for consumption by operators of content consumer devices, such as the content consumer device **14**. In some examples, the content creator device **12** may be operated by an individual user who would like to compress HOA coefficients **11**. Often, the content creator generates audio content in conjunction with video content. The content consumer device **14** may be operated by an individual. The content consumer device **14** may include an audio playback system **16**, which may refer to any form of audio playback system capable of rendering SHC for play back as multi-channel audio content.

The content creator device **12** includes an audio editing system **18**. The content creator device **12** obtain live recordings **7** in various formats (including directly as HOA coefficients) and audio objects **9**, which the content creator device **12** may edit using audio editing system **18**. A microphone **5** may capture the live recordings **7**. The content creator may, during the editing process, render HOA coefficients **11** from audio objects **9**, listening to the rendered speaker feeds in an attempt to identify various aspects of the soundfield that require further editing. The content creator device **12** may then edit HOA coefficients **11** (potentially indirectly through manipulation of different ones of the audio objects **9** from which the source HOA coefficients may be derived in the manner described above). The content creator device **12** may employ the audio editing system **18** to generate the HOA coefficients **11**. The audio editing system **18** represents any system capable of editing audio data and outputting the audio data as one or more source spherical harmonic coefficients.

When the editing process is complete, the content creator device **12** may generate a bitstream **21** based on the HOA coefficients **11**. That is, the content creator device **12** includes an audio encoding device **20** that represents a device configured to encode or otherwise compress HOA coefficients **11** in accordance with various aspects of the techniques described in this disclosure to generate the bitstream **21**. The audio encoding device **20** may generate the bitstream **21** for transmission, as one example, across a transmission channel, which may be a wired or wireless channel, a data storage device, or the like. The bitstream **21** may represent an encoded version of the HOA coefficients **11** and may include a primary bitstream and another side bitstream, which may be referred to as side channel information.

While shown in FIG. **2** as being directly transmitted to the content consumer device **14**, the content creator device **12** may output the bitstream **21** to an intermediate device positioned between the content creator device **12** and the content consumer device **14**. The intermediate device may store the bitstream **21** for later delivery to the content consumer device **14**, which may request the bitstream. The intermediate device may comprise a file server, a web server, a desktop computer, a laptop computer, a tablet computer, a mobile phone, a smart phone, or any other device capable of storing the bitstream **21** for later retrieval by an audio decoder. The intermediate device may reside in a content delivery network capable of streaming the bitstream **21** (and possibly in conjunction with transmitting a corresponding video data bitstream) to subscribers, such as the content consumer device **14**, requesting the bitstream **21**.

Alternatively, the content creator device **12** may store the bitstream **21** to a storage medium, such as a compact disc, a digital video disc, a high definition video disc or other storage media, most of which are capable of being read by a computer and therefore may be referred to as computer-readable storage media or non-transitory computer-readable storage media. In this context, the transmission channel may refer to the channels by which content stored to the mediums are transmitted (and may include retail stores and other store-based delivery mechanism). In any event, the techniques of this disclosure should not therefore be limited in this respect to the example of FIG. **2**.

As further shown in the example of FIG. **2**, the content consumer device **14** includes the audio playback system **16**. The audio playback system **16** may represent any audio playback system capable of playing back multi-channel audio data. The audio playback system **16** may include a number of different renderers **22**. The renderers **22** may each provide for a different form of rendering, where the different forms of rendering may include one or more of the various ways of performing vector-base amplitude panning (VBAP), and/or one or more of the various ways of performing soundfield synthesis. As used herein, "A and/or B" means "A or B", or both "A and B".

The audio playback system **16** may further include an audio decoding device **24**. The audio decoding device **24** may represent a device configured to decode HOA coeffi-

cients **11'** from the bitstream **21**, where the HOA coefficients **11'** may be similar to the HOA coefficients **11** but differ due to lossy operations (e.g., quantization) and/or transmission via the transmission channel. The audio playback system **16** may, after decoding the bitstream **21** to obtain the HOA coefficients **11'** and render the HOA coefficients **11'** to output loudspeaker feeds **25**. The loudspeaker feeds **25** may drive one or more loudspeakers (which are not shown in the example of FIG. **2** for ease of illustration purposes).

To select the appropriate renderer or, in some instances, generate an appropriate renderer, the audio playback system **16** may obtain loudspeaker information **13** indicative of a number of loudspeakers and/or a spatial geometry of the loudspeakers. In some instances, the audio playback system **16** may obtain the loudspeaker information **13** using a reference microphone and driving the loudspeakers in such a manner as to dynamically determine the loudspeaker information **13**. In other instances or in conjunction with the dynamic determination of the loudspeaker information **13**, the audio playback system **16** may prompt a user to interface with the audio playback system **16** and input the loudspeaker information **13**.

The audio playback system **16** may then select one of the audio renderers **22** based on the loudspeaker information **13**. In some instances, the audio playback system **16** may, when none of the audio renderers **22** are within some threshold similarity measure (in terms of the loudspeaker geometry) to the loudspeaker geometry specified in the loudspeaker information **13**, generate the one of audio renderers **22** based on the loudspeaker information **13**. The audio playback system **16** may, in some instances, generate one of the audio renderers **22** based on the loudspeaker information **13** without first attempting to select an existing one of the audio renderers **22**. One or more speakers **3** may then playback the rendered loudspeaker feeds **25**.

FIG. **3** is a block diagram illustrating, in more detail, one example of the audio encoding device **20** shown in the example of FIG. **2** that may perform various aspects of the techniques described in this disclosure. The audio encoding device **20** includes a content analysis unit **26**, a vector-based decomposition unit **27** and a directional-based decomposition unit **28**. Although described briefly below, more information regarding the audio encoding device **20** and the various aspects of compressing or otherwise encoding HOA coefficients is available in International Patent Application Publication No. WO 2014/194099, entitled "INTERPOLATION FOR DECOMPOSED REPRESENTATIONS OF A SOUND FIELD," filed 29 May 2014.

The content analysis unit **26** represents a unit configured to analyze the content of the HOA coefficients **11** to identify whether the HOA coefficients **11** represent content generated from a live recording or an audio object. The content analysis unit **26** may determine whether the HOA coefficients **11** were generated from a recording of an actual soundfield or from an artificial audio object. In some instances, when the framed HOA coefficients **11** were generated from a recording, the content analysis unit **26** passes the HOA coefficients **11** to the vector-based decomposition unit **27**. In some instances, when the framed HOA coefficients **11** were generated from a synthetic audio object, the content analysis unit **26** passes the HOA coefficients **11** to the directional-based synthesis unit **28**. The directional-based synthesis unit **28** may represent a unit configured to perform a directional-based synthesis of the HOA coefficients **11** to generate a directional-based bitstream **21**.

As shown in the example of FIG. **3**, the vector-based decomposition unit **27** may include a linear invertible transform (LIT) unit **30**, a parameter calculation unit **32**, a reorder unit **34**, a foreground selection unit **36**, an energy compensation unit **38**, a psychoacoustic audio coder unit **40**, a bitstream generation unit **42**, a soundfield analysis unit **44**, a coefficient reduction unit **46**, a background (BG) selection unit **48**, a spatio-temporal interpolation unit **50**, and a quantization unit **52**.

The linear invertible transform (LIT) unit **30** receives the HOA coefficients **11** in the form of HOA channels, each channel representative of a block or frame of a coefficient associated with a given order, sub-order of the spherical basis functions (which may be denoted as HOA[k], where k may denote the current frame or block of samples). The matrix of HOA coefficients **11** may have dimensions D: $M \times (N+1)^2$.

The LIT unit **30** may represent a unit configured to perform a form of analysis referred to as singular value decomposition. While described with respect to SVD, the techniques described in this disclosure may be performed with respect to any similar transformation or decomposition that provides for sets of linearly uncorrelated, energy compacted output. Also, reference to "sets" in this disclosure is generally intended to refer to non-zero sets unless specifically stated to the contrary and is not intended to refer to the classical mathematical definition of sets that includes the so-called "empty set." An alternative transformation may comprise a principal component analysis, which is often referred to as "PCA." Depending on the context, PCA may be referred to by a number of different names, such as discrete Karhunen-Loeve transform, the Hotelling transform, proper orthogonal decomposition (POD), and eigenvalue decomposition (EVD) to name a few examples. Properties of such operations that are conducive to the underlying goal of compressing audio data are 'energy compaction' and 'decorrelation' of the multichannel audio data.

In any event, assuming the LIT unit **30** performs a singular value decomposition (which, again, may be referred to as "SVD") for purposes of example, the LIT unit **30** may transform the HOA coefficients **11** into two or more sets of transformed HOA coefficients. The "sets" of transformed HOA coefficients may include vectors of transformed HOA coefficients. In the example of FIG. **3**, the LIT unit **30** may perform the SVD with respect to the HOA coefficients **11** to generate a so-called V matrix, an S matrix, and a U matrix. SVD, in linear algebra, may represent a factorization of a y-by-z real or complex matrix X (where X may represent multi-channel audio data, such as the HOA coefficients **11**) in the following form:

$$X = USV^*$$

U may represent a y-by-y real or complex unitary matrix, where the y columns of U are known as the left-singular vectors of the multi-channel audio data. S may represent a y-by-z rectangular diagonal matrix with non-negative real numbers on the diagonal, where the diagonal values of S are known as the singular values of the multi-channel audio data. V* (which may denote a conjugate transpose of V) may represent a z-by-z real or complex unitary matrix, where the z columns of V* are known as the right-singular vectors of the multi-channel audio data.

In some examples, the V* matrix in the SVD mathematical expression referenced above is denoted as the conjugate transpose of the V matrix to reflect that SVD may be applied to matrices comprising complex numbers. When applied to matrices comprising only real-numbers, the complex conjugate of the V matrix (or, in other words, the V* matrix) may be considered to be the transpose of the V matrix.

Below it is assumed, for ease of illustration purposes, that the HOA coefficients **11** comprise real-numbers with the result that the V matrix is output through SVD rather than the V* matrix. Moreover, while denoted as the V matrix in this disclosure, reference to the V matrix should be understood to refer to the transpose of the V matrix where appropriate. While assumed to be the V matrix, the techniques may be applied in a similar fashion to HOA coefficients **11** having complex coefficients, where the output of the SVD is the V* matrix. Accordingly, the techniques should not be limited in this respect to only provide for application of SVD to generate a V matrix, but may include application of SVD to HOA coefficients **11** having complex components to generate a V* matrix.

In this way, the LIT unit **30** may perform SVD with respect to the HOA coefficients **11** to output US[k] vectors **33** (which may represent a combined version of the S vectors and the U vectors) having dimensions D: $M\times(N+1)^2$, and V[k] vectors **35** having dimensions D: $(N+1)^2\times(N+1)^2$. Individual vector elements in the US[k] matrix may also be termed $X_{PS}(k)$ while individual vectors of the V[k] matrix may also be termed v(k).

An analysis of the U, S and V matrices may reveal that the matrices carry or represent spatial and temporal characteristics of the underlying soundfield represented above by X. Each of the N vectors in U (of length M samples) may represent normalized separated audio signals as a function of time (for the time period represented by M samples), that are orthogonal to each other and that have been decoupled from any spatial characteristics (which may also be referred to as directional information). The spatial characteristics, representing spatial shape and position (r, theta, phi) may instead be represented by individual i^{th} vectors, $v^{(i)}(k)$, in the V matrix (each of length $(N+1)^2$). The individual elements of each of $v^{(i)}(k)$ vectors may represent an HOA coefficient describing the shape (including width) and position of the soundfield for an associated audio object. Both the vectors in the U matrix and the V matrix are normalized such that their root-mean-square energies are equal to unity. The energy of the audio signals in U are thus represented by the diagonal elements in S. Multiplying U and S to form US[k] (with individual vector elements $X_{PS}(k)$), thus represent the audio signal with energies. The ability of the SVD decomposition to decouple the audio time-signals (in U), their energies (in S) and their spatial characteristics (in V) may support various aspects of the techniques described in this disclosure. Further, the model of synthesizing the underlying HOA[k] coefficients, X, by a vector multiplication of US[k] and V[k] gives rise the term "vector-based decomposition," which is used throughout this document.

Although described as being performed directly with respect to the HOA coefficients **11**, the LIT unit **30** may apply the linear invertible transform to derivatives of the HOA coefficients **11**. For example, the LIT unit **30** may apply SVD with respect to a power spectral density matrix derived from the HOA coefficients **11**. By performing SVD with respect to the power spectral density (PSD) of the HOA coefficients rather than the coefficients themselves, the LIT unit **30** may potentially reduce the computational complexity of performing the SVD in terms of one or more of processor cycles and storage space, while achieving the same source audio encoding efficiency as if the SVD were applied directly to the HOA coefficients.

The parameter calculation unit **32** represents a unit configured to calculate various parameters, such as a correlation parameter (R), directional properties parameters (θ, ω, r), and an energy property (e). Each of the parameters for the current frame may be denoted as R[k], θ[k], φ[k], r[k] and e[k]. The parameter calculation unit **32** may perform an energy analysis and/or correlation (or so-called cross-correlation) with respect to the US[k] vectors **33** to identify the parameters. The parameter calculation unit **32** may also determine the parameters for the previous frame, where the previous frame parameters may be denoted R[k−1], θ[k−1], φ[k−1], r[k−1] and e[k−1], based on the previous frame of US[k−1] vector and V[k−1] vectors. The parameter calculation unit **32** may output the current parameters **37** and the previous parameters **39** to reorder unit **34**.

The parameters calculated by the parameter calculation unit **32** may be used by the reorder unit **34** to re-order the audio objects to represent their natural evaluation or continuity over time. The reorder unit **34** may compare each of the parameters **37** from the first US[k] vectors **33** turn-wise against each of the parameters **39** for the second US[k−1] vectors **33**. The reorder unit **34** may reorder (using, as one example, a Hungarian algorithm) the various vectors within the US[k] matrix **33** and the V[k] matrix **35** based on the current parameters **37** and the previous parameters **39** to output a reordered US[k] matrix **33'** (which may be denoted mathematically as $\overline{US}[k]$) and a reordered V[k] matrix **35'** (which may be denoted mathematically as $\overline{V}[k]$) to a foreground sound (or predominant sound—PS) selection unit **36** ("foreground selection unit **36**") and an energy compensation unit **38**.

The soundfield analysis unit **44** may represent a unit configured to perform a soundfield analysis with respect to the HOA coefficients **11** so as to potentially achieve a target bitrate **41**. The soundfield analysis unit **44** may, based on the analysis and/or on a received target bitrate **41**, determine the total number of psychoacoustic coder instantiations (which may be a function of the total number of ambient or background channels (BG_{TOT}) and the number of foreground channels or, in other words, predominant channels. The total number of psychoacoustic coder instantiations can be denoted as numHOATransportChannels.

The soundfield analysis unit **44** may also determine, again to potentially achieve the target bitrate **41**, the total number of foreground channels (nFG) **45**, the minimum order of the background (or, in other words, ambient) soundfield (N_{BG} or, alternatively, MinAmbHOAorder), the corresponding number of actual channels representative of the minimum order of background soundfield (nBGa=(MinAmbHOAorder+1)^2), and indices (i) of additional BG HOA channels to send (which may collectively be denoted as background channel information **43** in the example of FIG. **3**). The background channel information **42** may also be referred to as ambient channel information **43**. Each of the channels that remains from numHOATransportChannels-nBGa, may either be an "additional background/ambient channel", an "active vector-based predominant channel", an "active directional based predominant signal" or "completely inactive". In one aspect, the channel types may be indicated (as a "ChannelType") syntax element by two bits (e.g. 00: directional based signal; 01: vector-based predominant signal; 10: additional ambient signal; 11: inactive signal). The total number of background or ambient signals, nBGa, may be given by (MinAmbHOAorder+1)^2+the number of times the index **10** (in the above example) appears as a channel type in the bitstream for that frame.

The soundfield analysis unit **44** may select the number of background (or, in other words, ambient) channels and the number of foreground (or, in other words, predominant) channels based on the target bitrate **41**, selecting more background and/or foreground channels when the target

bitrate **41** is relatively higher (e.g., when the target bitrate **41** equals or is greater than 512 Kbps). In one aspect, the numHOATransportChannels may be set to 8 while the MinAmbHOAorder may be set to 1 in the header section of the bitstream. In this scenario, at every frame, four channels may be dedicated to represent the background or ambient portion of the soundfield while the other 4 channels can, on a frame-by-frame basis vary on the type of channel—e.g., either used as an additional background/ambient channel or a foreground/predominant channel. The foreground/predominant signals can be one of either vector-based or directional based signals, as described above.

In some instances, the total number of vector-based predominant signals for a frame, may be given by the number of times the ChannelType index is 01 in the bitstream of that frame. In the above aspect, for every additional background/ambient channel (e.g., corresponding to a ChannelType of 10), corresponding information of which of the possible HOA coefficients (beyond the first four) may be represented in that channel. The information, for fourth order HOA content, may be an index to indicate the HOA coefficients **5-25**. The first four ambient HOA coefficients **1-4** may be sent all the time when minAmbHOAorder is set to 1, hence the audio encoding device may only need to indicate one of the additional ambient HOA coefficient having an index of 5-25. The information could thus be sent using a 5 bits syntax element (for 4^{th} order content), which may be denoted as "CodedAmbCoeffIdx." In any event, the soundfield analysis unit **44** outputs the background channel information **43**, the US[k] vectors **33**, and the V[k] vectors **35** to one or more other components of vector-based synthesis unit **27**B, such as the BG selection unit **48**B.

BG selection unit **48** may represent a unit configured to determine background or ambient V_{BG}[k] vectors $\mathbf{35}_{BG}$ based on the background channel information (e.g., the background soundfield (N_{BG}) and the number (nBGa) and the indices (i) of additional BG HOA channels to send). For example, when N_{BG} equals one, the background selection unit **48** may select the V[k] vectors **35** for each sample of the audio frame having an order equal to or less than one as the V_{BG}[k] vectors $\mathbf{35}_{BG}$. The background selection unit **48** may, in this example, then select the V[k] vectors **35** having an index identified by one of the indices (i) as additional V_{BG}[k] vectors $\mathbf{35}_{BG}$, where the nBGa is provided to the bitstream generation unit **42** to be specified in the bitstream **21** so as to enable the audio decoding device, such as the audio decoding device **24** shown in the example of FIG. **4**, to parse the BG HOA coefficients **47** from the bitstream **21**. The background selection unit **48** may then output the V_{BG}[k] vectors $\mathbf{35}_{BG}$ to one or more other components of crossfade unit **66**, such as energy compensation unit **38**. The V_{BG} [k] vectors $\mathbf{35}_{BG}$ may have dimensions D: $[(N_{BG}+1)^2+nBGa]\times(N+1)^2$. In some examples, BG selection unit **48** may also output the US[k] vectors **33** to one or more other components of crossfade unit **66**, such as energy compensation unit **38**.

Energy compensation unit **38** may represent a unit configured to perform energy compensation with respect to the V_{BG}[k] vectors $\mathbf{35}_{BG}$ to compensate for energy loss due to removal of various ones of the V[k] vectors **35** by the background selection unit **48**. The energy compensation unit **38** may perform an energy analysis with respect to one or more of the reordered US[k] matrix **33'**, the reordered V[k] matrix **35'**, the nFG signals **49**, the foreground V[k] vectors $\mathbf{51}_k$ and the V_{BG}[k] vectors $\mathbf{35}_{BG}$ and then perform energy compensation based on this energy analysis to generate energy compensated V_{BG}[k] vectors $\mathbf{35}_{BG}$'. The energy compensation unit **38** may output the energy compensated V_{BG}[k] vectors $\mathbf{35}_{BG}$' to one or more other components of vector-based synthesis unit **27**, such as matrix math unit **64**. In some examples, energy compensation unit **38** may also output the US[k] vectors **33** to one or more other components of crossfade unit **66**, such as matrix math unit **64**.

Matrix math unit **64** may represent a unit configured to perform any variety of operations on one or more matrices. In the example of FIG. **3**, matrix math unit **64** may be configured to multiply the US[k] vectors **33** by energy compensated V_{BG}[k] vectors $\mathbf{35}_{BG}$' to obtain energy compensated ambient HOA coefficients **47'**. Matrix math unit **64** may provide the determined energy compensated ambient HOA coefficients **47'** to one or more other components of vector-based synthesis unit **27**, such as crossfade unit **66**. The energy compensated ambient HOA coefficients **47'** may have dimensions D: $M\times[(N_{BG}+1)^2+nBGa]$.

Crossfade unit **66** may represent a unit configured to perform crossfading between signals. For instance, crossfade unit **66** may crossfade between the energy compensated ambient HOA coefficients **47'** of frame k and the energy compensated ambient HOA coefficients **47'** of a previous frame k−1 to determine crossfaded energy compensated ambient HOA coefficients **47"** for frame k. Crossfade unit **66** may output the determine crossfaded energy compensated ambient HOA coefficients **47"** for frame k to one or more other components of vector-based synthesis unit **27**, such as psychoacoustic audio coder unit **40**.

In some examples, crossfade unit **66** may crossfade between the energy compensated ambient HOA coefficients **47'** of frame k and the energy compensated ambient HOA coefficients **47'** of a previous frame k−1 by modifying a portion of the energy compensated ambient HOA coefficients **47'** of frame k based on a portion of the energy compensated ambient HOA coefficients **47'** of frame k−1. In some examples, crossfade unit **66** may remove a portion of the coefficients when determining crossfaded energy compensated ambient HOA coefficients **47"**. Additional details of crossfade unit **66** are provided below with reference to FIG. **14**.

The foreground selection unit **36** may represent a unit configured to select the reordered US[k] matrix **33'** and the reordered V[k] matrix **35'** that represent foreground or distinct components of the soundfield based on nFG **45** (which may represent a one or more indices identifying the foreground vectors). The foreground selection unit **36** may output nFG signals **49** (which may be denoted as a reordered $US[k]_{1,\ldots,nFG}$ **49**, $FG_{1,\ldots,nfG}[k]$ **49**, or $X_{PS}^{(1\ldots nFG)}(k)$ **49**) to the psychoacoustic audio coder unit **40**, where the nFG signals **49** may have dimensions D: M×nFG and each represent mono-audio objects. The foreground selection unit **36** may also output the reordered V[k] matrix **35'** (or $v^{(1\ldots nFG)}(k)$ **35'**) corresponding to foreground components of the soundfield to the spatio-temporal interpolation unit **50**, where a subset of the reordered V[k] matrix **35'** corresponding to the foreground components may be denoted as foreground V[k] matrix $\mathbf{51}_k$ (which may be mathematically denoted as, $\overline{V}_{1,\ldots,nFG}[k]$) having dimensions D: $(N+1)^2\times$ nFG.

The spatio-temporal interpolation unit **50** may represent a unit configured to receive the foreground V[k] vectors $\mathbf{51}_k$ for the k'th frame and the foreground V[k−1] vectors $\mathbf{51}_{k-1}$ for the previous frame (hence the k−1 notation) and perform spatio-temporal interpolation to generate interpolated foreground V[k] vectors. The spatio-temporal interpolation unit **50** may recombine the nFG signals **49** with the foreground V[k] vectors $\mathbf{51}_k$ to recover reordered foreground HOA

coefficients. The spatio-temporal interpolation unit **50** may then divide the reordered foreground HOA coefficients by the interpolated V[k] vectors to generate interpolated nFG signals **49'**. The spatio-temporal interpolation unit **50** may also output those of the foreground V[k] vectors $\mathbf{51}_k$ that were used to generate the interpolated foreground V[k] vectors so that an audio decoding device, such as the audio decoding device **24**, may generate the interpolated foreground V[k] vectors and thereby recover the foreground V[k] vectors $\mathbf{51}_k$. Those of the foreground V[k] vectors $\mathbf{51}_k$ used to generate the interpolated foreground V[k] vectors are denoted as the remaining foreground V[k] vectors **53**. In order to ensure that the same V[k] and V[k−1] are used at the encoder and decoder (to create the interpolated vectors V[k]) quantized/dequantized versions of these may be used at the encoder and decoder.

In this respect, the spatio-temporal interpolation unit **50** may represent a unit that interpolates a first portion of a first audio frame from some other portions of the first audio frame and a second temporally subsequent or preceding audio frame. In some examples, the portions may be denoted as sub-frames, where interpolation as performed with respect to sub-frames is described in more detail below with respect to FIGS. **45-46**E. In other examples, the spatio-temporal interpolation unit **50** may operate with respect to some last number of samples of the previous frame and some first number of samples of the subsequent frame, as described in more detail with respect to FIGS. **37-39**. The spatio-temporal interpolation unit **50** may, in performing this interpolation, reduce the number of samples of the foreground V[k] vectors $\mathbf{51}_k$ that are required to be specified in the bitstream **21**, as only those of the foreground V[k] vectors $\mathbf{51}_k$ that are used to generate the interpolated V[k] vectors represent a subset of the foreground V[k] vectors $\mathbf{51}_k$. That is, in order to potentially make compression of the HOA coefficients **11** more efficient (by reducing the number of the foreground V[k] vectors $\mathbf{51}_k$ that are specified in the bitstream **21**), various aspects of the techniques described in this disclosure may provide for interpolation of one or more portions of the first audio frame, where each of the portions may represent decomposed versions of the HOA coefficients **11**.

The spatio-temporal interpolation may result in a number of benefits. First, the nFG signals **49** may not be continuous from frame to frame due to the block-wise nature in which the SVD or other LIT is performed. In other words, given that the LIT unit **30** applies the SVD on a frame-by-frame basis, certain discontinuities may exist in the resulting transformed HOA coefficients as evidence for example by the unordered nature of the US[k] matrix **33** and V[k] matrix **35**. By performing this interpolation, the discontinuity may be reduced given that interpolation may have a smoothing effect that potentially reduces any artifacts introduced due to frame boundaries (or, in other words, segmentation of the HOA coefficients **11** into frames). Using the foreground V[k] vectors $\mathbf{51}_k$ to perform this interpolation and then generating the interpolated nFG signals **49'** based on the interpolated foreground V[k] vectors $\mathbf{51}_k$ from the recovered reordered HOA coefficients may smooth at least some effects due to the frame-by-frame operation as well as due to reordering the nFG signals **49**.

In operation, the spatio-temporal interpolation unit **50** may interpolate one or more sub-frames of a first audio frame from a first decomposition, e.g., foreground V[k] vectors $\mathbf{51}_k$, of a portion of a first plurality of the HOA coefficients **11** included in the first frame and a second decomposition, e.g., foreground V[k] vectors $\mathbf{51}_{k-1}$, of a portion of a second plurality of the HOA coefficients **11** included in a second frame to generate decomposed interpolated spherical harmonic coefficients for the one or more sub-frames.

In some examples, the first decomposition comprises the first foreground V[k] vectors $\mathbf{51}_k$ representative of right-singular vectors of the portion of the HOA coefficients **11**. Likewise, in some examples, the second decomposition comprises the second foreground V[k] vectors $\mathbf{51}_k$ representative of right-singular vectors of the portion of the HOA coefficients **11**.

In other words, spherical harmonics-based 3D audio may be a parametric representation of the 3D pressure field in terms of orthogonal basis functions on a sphere. The higher the order N of the representation, the potentially higher the spatial resolution, and often the larger the number of spherical harmonics (SH) coefficients (for a total of $(N+1)^2$ coefficients). For many applications, a bandwidth compression of the coefficients may be required for being able to transmit and store the coefficients efficiently. This techniques directed in this disclosure may provide a frame-based, dimensionality reduction process using Singular Value Decomposition (SVD). The SVD analysis may decompose each frame of coefficients into three matrices U, S and V. In some examples, the techniques may handle some of the vectors in US[k] matrix as foreground components of the underlying soundfield. However, when handled in this manner, these vectors (in U S[k] matrix) are discontinuous from frame to frame—even though they represent the same distinct audio component. These discontinuities may lead to significant artifacts when the components are fed through transform-audio-coders.

The techniques described in this disclosure may address this discontinuity. That is, the techniques may be based on the observation that the V matrix can be interpreted as orthogonal spatial axes in the Spherical Harmonics domain. The U[k] matrix may represent a projection of the Spherical Harmonics (HOA) data in terms of those basis functions, where the discontinuity can be attributed to orthogonal spatial axis (V[k]) that change every frame—and are therefore discontinuous themselves. This is unlike similar decomposition, such as the Fourier Transform, where the basis functions are, in some examples, constant from frame to frame. In these terms, the SVD may be considered of as a matching pursuit algorithm. The techniques described in this disclosure may enable the spatio-temporal interpolation unit **50** to maintain the continuity between the basis functions (V[k]) from frame to frame—by interpolating between them.

As noted above, the interpolation may be performed with respect to samples. This case is generalized in the above description when the subframes comprise a single set of samples. In both the case of interpolation over samples and over subframes, the interpolation operation may take the form of the following equation:

$$\overline{v(l)}=w(l)v(k)+(1-w(l))v(k-1).$$

In this above equation, the interpolation may be performed with respect to the single V-vector v(k) from the single V-vector v(k−1), which in one embodiment could represent V-vectors from adjacent frames k and k−1. In the above equation, l, represents the resolution over which the interpolation is being carried out, where l may indicate a integer sample and l=1, . . . , T (where T is the length of samples over which the interpolation is being carried out and over which the output interpolated vectors, $\overline{v(l)}$ are required and also indicates that the output of this process produces l of

these vectors). Alternatively, l could indicate subframes consisting of multiple samples. When, for example, a frame is divided into four subframes, l may comprise values of 1, 2, 3 and 4, for each one of the subframes. The value of l may be signaled as a field termed "CodedSpatialInterpolationTime" through a bitstream—so that the interpolation operation may be replicated in the decoder. The w(l) may comprise values of the interpolation weights. When the interpolation is linear, w(l) may vary linearly and monotonically between 0 and 1, as a function of l. In other instances, w(l) may vary between 0 and 1 in a non-linear but monotonic fashion (such as a quarter cycle of a raised cosine) as a function of l. The function, w(l), may be indexed between a few different possibilities of functions and signaled in the bitstream as a field termed "SpatialInterpolationMethod" such that the identical interpolation operation may be replicated by the decoder. When w(l) is a value close to 0, the output, $\overline{v(l)}$ may be highly weighted or influenced by v(k−1). Whereas when w(l) is a value close to 1, it ensures that the output, $\overline{v(l)}$, is highly weighted or influenced by v(k−1).

The coefficient reduction unit **46** may represent a unit configured to perform coefficient reduction with respect to the remaining foreground V[k] vectors **53** based on the background channel information **43** to output reduced foreground V[k] vectors **55** to the quantization unit **52**. The reduced foreground V[k] vectors **55** may have dimensions D: $[(N+1)^2-(N_{BG}+1)^2-BG_{TOT}]\times nFG$. The coefficient reduction unit **46** may, in this respect, represent a unit configured to reduce the number of coefficients in the remaining foreground V[k] vectors **53**. In other words, coefficient reduction unit **46** may represent a unit configured to eliminate the coefficients in the foreground V[k] vectors (that form the remaining foreground V[k] vectors **53**) having little to no directional information. In some examples, the coefficients of the distinct or, in other words, foreground V[k] vectors corresponding to a first and zero order basis functions (which may be denoted as N_{BG}) provide little directional information and therefore can be removed from the foreground V-vectors (through a process that may be referred to as "coefficient reduction"). In this example, greater flexibility may be provided to not only identify the coefficients that correspond N_{BG} but to identify additional HOA channels (which may be denoted by the variable TotalOfAddAmbHOAChan) from the set of $[(N_{BG}+1)^2+1, (N+1)^2]$.

The quantization unit **52** may represent a unit configured to perform any form of quantization to compress the reduced foreground V[k] vectors **55** to generate coded foreground V[k] vectors **57**, outputting the coded foreground V[k] vectors **57** to the bitstream generation unit **42**. In operation, the quantization unit **52** may represent a unit configured to compress a spatial component of the soundfield, i.e., one or more of the reduced foreground V[k] vectors **55** in this example. The quantization unit **52** may perform any one of the following 12 quantization modes, as indicated by a quantization mode syntax element denoted "NbitsQ":

NbitsQ value	Type of Quantization Mode
0-3:	Reserved
4:	Vector Quantization
5:	Scalar Quantization without Huffman Coding
6:	6-bit Scalar Quantization with Huffman Coding
7:	7-bit Scalar Quantization with Huffman Coding
8:	8-bit Scalar Quantization with Huffman Coding
. . .	. . .
16:	16-bit Scalar Quantization with Huffman Coding

The quantization unit **52** may also perform predicted versions of any of the foregoing types of quantization modes, where a difference is determined between an element of (or a weight when vector quantization is performed) of the V-vector of a previous frame and the element (or weight when vector quantization is performed) of the V-vector of a current frame is determined. The quantization unit **52** may then quantize the difference between the elements or weights of the current frame and previous frame rather than the value of the element of the V-vector of the current frame itself.

The quantization unit **52** may perform multiple forms of quantization with respect to each of the reduced foreground V[k] vectors **55** to obtain multiple coded versions of the reduced foreground V[k] vectors **55**. The quantization unit **52** may select the one of the coded versions of the reduced foreground V[k] vectors **55** as the coded foreground V[k] vector **57**. The quantization unit **52** may, in other words, select one of the non-predicted vector-quantized V-vector, predicted vector-quantized V-vector, the non-Huffman-coded scalar-quantized V-vector, and the Huffman-coded scalar-quantized V-vector to use as the output switched-quantized V-vector based on any combination of the criteria discussed in this disclosure. In some examples, the quantization unit **52** may select a quantization mode from a set of quantization modes that includes a vector quantization mode and one or more scalar quantization modes, and quantize an input V-vector based on (or according to) the selected mode. The quantization unit **52** may then provide the selected one of the non-predicted vector-quantized V-vector (e.g., in terms of weight values or bits indicative thereof), predicted vector-quantized V-vector (e.g., in terms of error values or bits indicative thereof), the non-Huffman-coded scalar-quantized V-vector and the Huffman-coded scalar-quantized V-vector to the bitstream generation unit **52** as the coded foreground V[k] vectors **57**. The quantization unit **52** may also provide the syntax elements indicative of the quantization mode (e.g., the NbitsQ syntax element) and any other syntax elements used to dequantize or otherwise reconstruct the V-vector.

The psychoacoustic audio coder unit **40** included within the audio encoding device **20** may represent multiple instances of a psychoacoustic audio coder, each of which is used to encode a different audio object or HOA channel of each of the energy compensated ambient HOA coefficients **47'** and the interpolated nFG signals **49'** to generate encoded ambient HOA coefficients **59** and encoded nFG signals **61**. The psychoacoustic audio coder unit **40** may output the encoded ambient HOA coefficients **59** and the encoded nFG signals **61** to the bitstream generation unit **42**.

The bitstream generation unit **42** included within the audio encoding device **20** represents a unit that formats data to conform to a known format (which may refer to a format known by a decoding device), thereby generating the vector-based bitstream **21**. The bitstream **21** may, in other words, represent encoded audio data, having been encoded in the manner described above. The bitstream generation unit **42** may represent a multiplexer in some examples, which may receive the coded foreground V[k] vectors **57**, the encoded ambient HOA coefficients **59**, the encoded nFG signals **61** and the background channel information **43**. The bitstream generation unit **42** may then generate a bitstream **21** based on the coded foreground V[k] vectors **57**, the encoded ambient HOA coefficients **59**, the encoded nFG signals **61** and the background channel information **43**. In this way, the bitstream generation unit **42** may thereby specify the vectors **57** in the bitstream **21** to obtain the bitstream **21** as described below in more detail with respect to the example of FIG. **7**.

The bitstream **21** may include a primary or main bitstream and one or more side channel bitstreams.

Although not shown in the example of FIG. **3**, the audio encoding device **20** may also include a bitstream output unit that switches the bitstream output from the audio encoding device **20** (e.g., between the directional-based bitstream **21** and the vector-based bitstream **21**) based on whether a current frame is to be encoded using the directional-based synthesis or the vector-based synthesis. The bitstream output unit may perform the switch based on the syntax element output by the content analysis unit **26** indicating whether a directional-based synthesis was performed (as a result of detecting that the HOA coefficients **11** were generated from a synthetic audio object) or a vector-based synthesis was performed (as a result of detecting that the HOA coefficients were recorded). The bitstream output unit may specify the correct header syntax to indicate the switch or current encoding used for the current frame along with the respective one of the bitstreams **21**.

Moreover, as noted above, the soundfield analysis unit **44** may identify BG_{TOT} ambient HOA coefficients **47**, which may change on a frame-by-frame basis (although at times BG_{TOT} may remain constant or the same across two or more adjacent (in time) frames). The change in BG_{TOT} may result in changes to the coefficients expressed in the reduced foreground V[k] vectors **55**. The change in BG_{TOT} may result in background HOA coefficients (which may also be referred to as "ambient HOA coefficients") that change on a frame-by-frame basis (although, again, at times BG_{TOT} may remain constant or the same across two or more adjacent (in time) frames). The changes often result in a change of energy for the aspects of the sound field represented by the addition or removal of the additional ambient HOA coefficients and the corresponding removal of coefficients from or addition of coefficients to the reduced foreground V[k] vectors **55**.

As a result, the soundfield analysis unit **44** may further determine when the ambient HOA coefficients change from frame to frame and generate a flag or other syntax element indicative of the change to the ambient HOA coefficient in terms of being used to represent the ambient components of the sound field (where the change may also be referred to as a "transition" of the ambient HOA coefficient or as a "transition" of the ambient HOA coefficient). In particular, the coefficient reduction unit **46** may generate the flag (which may be denoted as an AmbCoeffTransition flag or an AmbCoeffIdxTransition flag), providing the flag to the bitstream generation unit **42** so that the flag may be included in the bitstream **21** (possibly as part of side channel information).

The coefficient reduction unit **46** may, in addition to specifying the ambient coefficient transition flag, also modify how the reduced foreground V[k] vectors **55** are generated. In one example, upon determining that one of the ambient HOA ambient coefficients is in transition during the current frame, the coefficient reduction unit **46** may specify, a vector coefficient (which may also be referred to as a "vector element" or "element") for each of the V-vectors of the reduced foreground V[k] vectors **55** that corresponds to the ambient HOA coefficient in transition. Again, the ambient HOA coefficient in transition may add or remove from the BG_{TOT} total number of background coefficients. Therefore, the resulting change in the total number of background coefficients affects whether the ambient HOA coefficient is included or not included in the bitstream, and whether the corresponding element of the V-vectors are included for the V-vectors specified in the bitstream in the second and third configuration modes described above. More information regarding how the coefficient reduction unit **46** may specify the reduced foreground V[k] vectors **55** to overcome the changes in energy is provided in U.S. application Ser. No. 14/594,533, entitled "TRANSITIONING OF AMBIENT HIGHER ORDER AMBISONIC COEFFICIENTS," filed Jan. 12, 2015.

FIG. **14** is a block diagram illustrating, in more detail, the crossfade unit **66** of the audio encoding device **20** shown in the example of FIG. **3**. The crossfade unit **66** may include a mixer unit **70**, a framing unit **71**, and a delay unit **72**. FIG. **14** illustrates only one example of the crossfade unit **66** and other configurations are possible. For instance, the framing unit **71** may be positioned prior to the mixer unit **70** such that a third portion **75** is removed before the energy compensated ambient HOA coefficients **47'** are received by the mixer unit **70**.

The mixer unit **70** may represent a unit configured to combine a plurality of signals into a single signal. For instance, the mixer unit **70** may combine a first signal with a second signal generate a modified signal. The mixer unit **70** may combine the first signal with the second signal by fading the first signal in while fading the second signal out. The mixer unit **70** may apply any variety of functions to fade the portions in and out. As one example, the mixer unit **70** may apply a linear function to fade the first signal in and apply a linear function to fade the second signal out. As another example, the mixer unit **70** may apply an exponential function to fade the first signal in and apply an exponential function to fade the second signal out. In some examples, the mixer unit **70** may apply different functions to the signals. For instance, the mixer unit **70** may apply a linear function to fade the first signal in and apply an exponential to fade the second signal out. In some examples, the mixer unit **70** may fade a signal in or out by fading a portion of the signal in or out. In any case, mixer unit may output the modified signal to one or more other components of the crossfade unit **66**, such as the framing unit **71**.

The framing unit **71** may represent a unit configured to frame an input signal to fit one or more particular dimensions. In some examples, such as where one or more of the dimensions of the input signal are larger than one or more of the particular dimensions, the framing unit **71** may generate a framed output signal by removing a portion of the input signal, e.g., the portion that exceeds the particular dimensions. For instance, where the particular dimensions are 1024 by 4 and the input signal has dimensions of 1280 by 4, the framing unit **71** may generate the framed output signal by removing a 256 by 4 portion of the input signal. In some examples, the framing unit **71** may output the framed output signal to one or more other components of the audio encoding device **20**, such as the psychoacoustic audio coder unit **40** of FIG. **3**. In some examples, framing unit **71** may output the removed portion of the input signal to one or more other components of the crossfade unit **66**, such as the delay unit **72**.

The delay unit **72** may represent a unit configured to store a signal for later use. For instance, the delay unit **72** may be configured to store, at a first time, a first signal and output, at a second later time, the first signal. In this way, the delay unit **72** may operate as a first-in first-out (FIFO) buffer. The delay unit **72** may output, at the second later time, the first signal to one or more other components of the crossfade unit **66**, such as the mixer unit **70**.

As discussed above, the crossfade unit **66** may receive the energy compensated ambient HOA coefficients **47'** of a current frame (e.g., frame k), crossfade the energy compen-

sated ambient HOA coefficients **47'** of the current frame with the energy compensated ambient HOA coefficients **47'** of a previous frame, and output the crossfaded energy compensated ambient HOA coefficients **47"**. As illustrated in FIG. **14**, the energy compensated ambient HOA coefficients **47'** may include a first portion **73**, a second portion **74**, and a third portion **75**.

In accordance with one or more techniques of this disclosure, the mixer unit **70** of the crossfade unit **66** may combine (e.g., crossfade between) the first portion **73** of the energy compensated ambient HOA coefficients **47'** of the current frame and the third portion **76** of the energy compensated ambient HOA coefficients **47'** of the previous frame to generate intermediate crossfaded energy compensated ambient HOA coefficients **77**. The mixer unit **70** may output the generated intermediate crossfaded energy compensated ambient HOA coefficients **77** to the framing unit **71**. As the mixer unit **70** utilizes the third portion **76** of the energy compensated ambient HOA coefficients **47'** of the previous frame, in this example, it can be assumed that the crossfade unit **66** was in operation prior to processing the current frame. As such, as opposed to separately crossfading the US matrix of the current frame with the US matrix of the previous frame and the V matrix of the current frame with the V matrix of the previous frame, the mixer unit **70** may crossfade in the energy compensated domain. In this way, techniques according to the disclosure may reduce the computational load, power consumption, and/or complexity of the crossfade unit **66**.

The framing unit **71** may determine the crossfaded energy compensated ambient HOA coefficients **47"** by removing the third portion **75** from the intermediate crossfaded energy compensated ambient HOA coefficients **77** if the dimensions of the intermediate crossfaded energy compensated ambient HOA coefficients **77** exceed the dimensions of the current frame. For instance, where the dimensions for the current frame are 1024 by 4 and the dimension of the intermediate crossfaded energy compensated ambient HOA coefficients **77** are 1280 by 4, the framing unit **71** may determine the crossfaded energy compensated ambient HOA coefficients **47"** by removing the third portion **75** (e.g., a 256 by 4 portion) from the intermediate crossfaded energy compensated ambient HOA coefficients **77**. The framing unit **71** may output the third portion **75** to delay unit **72** for future use (e.g., by the mixer unit **70** when crossfading the energy compensated ambient HOA coefficients **47'** of a subsequent frame). The framing unit **71** may output the determined crossfaded energy compensated ambient HOA coefficients **47"** to the psychoacoustic audio coder unit **40** of FIG. **3**. In this way, the crossfade unit **66** may smooth the transition between the previous frame and the current frame.

In some examples, the crossfade unit **66** may crossfade between any two sets of HOA coefficients. As one example, the crossfade unit **66** may crossfade between a first set of HOA coefficients and a second set of HOA coefficients. As another example, the crossfade unit **66** may crossfade between a current set of HOA coefficients and a previous set of HOA coefficients.

FIG. **4** is a block diagram illustrating the audio decoding device **24** of FIG. **2** in more detail. As shown in the example of FIG. **4** the audio decoding device **24** may include an extraction unit **72**, a directionality-based reconstruction unit **90** and a vector-based reconstruction unit **92**. Although described below, more information regarding the audio decoding device **24** and the various aspects of decompressing or otherwise decoding HOA coefficients is available in International Patent Application Publication No. WO 2014/194099, entitled "INTERPOLATION FOR DECOMPOSED REPRESENTATIONS OF A SOUND FIELD," filed 29 May 2014.

The extraction unit **72** may represent a unit configured to receive the bitstream **21** and extract the various encoded versions (e.g., a directional-based encoded version or a vector-based encoded version) of the HOA coefficients **11**. The extraction unit **72** may determine from the above noted syntax element indicative of whether the HOA coefficients **11** were encoded via the various direction-based or vector-based versions. When a directional-based encoding was performed, the extraction unit **72** may extract the directional-based version of the HOA coefficients **11** and the syntax elements associated with the encoded version (which is denoted as directional-based information **91** in the example of FIG. **4**), passing the directional based information **91** to the directional-based reconstruction unit **90**. The directional-based reconstruction unit **90** may represent a unit configured to reconstruct the HOA coefficients in the form of HOA coefficients **11'** based on the directional-based information **91**.

When the syntax element indicates that the HOA coefficients **11** were encoded using a vector-based synthesis, the extraction unit **72** may extract the coded foreground V[k] vectors **57** (which may include coded weights **57** and/or indices **63** or scalar quantized V-vectors), the encoded ambient HOA coefficients **59** and the corresponding audio objects **61** (which may also be referred to as the encoded nFG signals **61**). The audio objects **61** each correspond to one of the vectors **57**. The extraction unit **72** may pass the coded foreground V[k] vectors **57** to the V-vector reconstruction unit **74** and the encoded ambient HOA coefficients **59** along with the encoded nFG signals **61** to the psychoacoustic decoding unit **80**.

The V-vector reconstruction unit **74** may represent a unit configured to reconstruct the V-vectors from the encoded foreground V[k] vectors **57**. The V-vector reconstruction unit **74** may operate in a manner reciprocal to that of the quantization unit **52**.

The psychoacoustic decoding unit **80** may operate in a manner reciprocal to the psychoacoustic audio coder unit **40** shown in the example of FIG. **3** so as to decode the encoded ambient HOA coefficients **59** and the encoded nFG signals **61** and thereby generate energy compensated ambient HOA coefficients **47'** and the interpolated nFG signals **49'** (which may also be referred to as interpolated nFG audio objects **49'**). The psychoacoustic decoding unit **80** may pass the energy compensated ambient HOA coefficients **47'** to the fade unit **770** and the nFG signals **49'** to the foreground formulation unit **78**.

The spatio-temporal interpolation unit **76** may operate in a manner similar to that described above with respect to the spatio-temporal interpolation unit **50**. The spatio-temporal interpolation unit **76** may receive the reduced foreground V[k] vectors $\mathbf{55}_k$ and perform the spatio-temporal interpolation with respect to the foreground V[k] vectors $\mathbf{55}_k$ and the reduced foreground V[k−1] vectors $\mathbf{55}_{k-1}$ to generate interpolated foreground V[k] vectors $\mathbf{55}_k$". The spatio-temporal interpolation unit **76** may forward the interpolated foreground V[k] vectors $\mathbf{55}_k$" to the fade unit **770**.

The extraction unit **72** may also output a signal **757** indicative of when one of the ambient HOA coefficients is in transition to fade unit **770**, which may then determine which of the SHC_{BG} **47'** (where the SHC_{BG} **47'** may also be denoted as "ambient HOA channels **47**" or "ambient HOA coefficients **47'**) and the elements of the interpolated foreground V[k] vectors $\mathbf{55}_k$" are to be either faded-in or faded-out. In

some examples, the fade unit **770** may operate opposite with respect to each of the ambient HOA coefficients **47'** and the elements of the interpolated foreground V[k] vectors $\mathbf{55}_k$". That is, the fade unit **770** may perform a fade-in or fade-out, or both a fade-in or fade-out with respect to corresponding one of the ambient HOA coefficients **47'**, while performing a fade-in or fade-out or both a fade-in and a fade-out, with respect to the corresponding one of the elements of the interpolated foreground V[k] vectors $\mathbf{55}_k$". The fade unit **770** may output adjusted ambient HOA coefficients **47"** to the HOA coefficient formulation unit **82** and adjusted foreground V[k] vectors $\mathbf{55}_k$'" to the foreground formulation unit **78**. In this respect, the fade unit **770** represents a unit configured to perform a fade operation with respect to various aspects of the HOA coefficients or derivatives thereof, e.g., in the form of the ambient HOA coefficients **47'** and the elements of the interpolated foreground V[k] vectors $\mathbf{55}_k$".

The foreground formulation unit **78** may represent a unit configured to perform matrix multiplication with respect to the adjusted foreground V[k] vectors $\mathbf{55}_k$'" and the interpolated nFG signals **49'** to generate the foreground HOA coefficients **65**. In this respect, the foreground formulation unit **78** may combine the audio objects **49'** (which is another way by which to denote the interpolated nFG signals **49'**) with the vectors $\mathbf{55}_k$'" to reconstruct the foreground or, in other words, predominant aspects of the HOA coefficients **11'**. The foreground formulation unit **78** may perform a matrix multiplication of the interpolated nFG signals **49'** by the adjusted foreground V[k] vectors $\mathbf{55}_k$'".

The HOA coefficient formulation unit **82** may represent a unit configured to combine the foreground HOA coefficients **65** to the adjusted ambient HOA coefficients **47"** so as to obtain the HOA coefficients **11'**. The prime notation reflects that the HOA coefficients **11'** may be similar to but not the same as the HOA coefficients **11**. The differences between the HOA coefficients **11** and **11'** may result from loss due to transmission over a lossy transmission medium, quantization or other lossy operations.

FIG. **5** is a flowchart illustrating exemplary operation of an audio encoding device, such as the audio encoding device **20** shown in the example of FIG. **3**, in performing various aspects of the vector-based synthesis techniques described in this disclosure. Initially, the audio encoding device **20** receives the HOA coefficients **11** (**106**). The audio encoding device **20** may invoke the LIT unit **30**, which may apply a LIT with respect to the HOA coefficients to output transformed HOA coefficients (e.g., in the case of SVD, the transformed HOA coefficients may comprise the US[k] vectors **33** and the V[k] vectors **35**) (**107**).

The audio encoding device **20** may next invoke the parameter calculation unit **32** to perform the above described analysis with respect to any combination of the US[k] vectors **33**, US[k−1] vectors **33**, the V[k] and/or V[k−1] vectors **35** to identify various parameters in the manner described above. That is, the parameter calculation unit **32** may determine at least one parameter based on an analysis of the transformed HOA coefficients **33/35** (**108**).

The audio encoding device **20** may then invoke the reorder unit **34**, which may reorder the transformed HOA coefficients (which, again in the context of SVD, may refer to the US[k] vectors **33** and the V[k] vectors **35**) based on the parameter to generate reordered transformed HOA coefficients **33'/35'** (or, in other words, the US[k] vectors **33'** and the V[k] vectors **35'**), as described above (**109**). The audio encoding device **20** may, during any of the foregoing operations or subsequent operations, also invoke the soundfield analysis unit **44**. The soundfield analysis unit **44** may, as described above, perform a soundfield analysis with respect to the HOA coefficients **11** and/or the transformed HOA coefficients **33/35** to determine the total number of foreground channels (nFG) **45**, the order of the background soundfield (N_{BG}) and the number (nBGa) and indices (i) of additional BG HOA channels to send (which may collectively be denoted as background channel information **43** in the example of FIG. **3**) (**109**).

The audio encoding device **20** may also invoke the background selection unit **48**. The background selection unit **48** may determine background or ambient HOA coefficients **47** based on the background channel information **43** (**110**). The audio encoding device **20** may further invoke the foreground selection unit **36**, which may select the reordered US[k] vectors **33'** and the reordered V[k] vectors **35'** that represent foreground or distinct components of the soundfield based on nFG **45** (which may represent a one or more indices identifying the foreground vectors) (**112**).

The audio encoding device **20** may invoke the energy compensation unit **38**. The energy compensation unit **38** may perform energy compensation with respect to the ambient HOA coefficients **47** to compensate for energy loss due to removal of various ones of the HOA coefficients by the background selection unit **48** and crossfade energy compensated ambient HOA coefficients **47'** in the manner described above (**114**).

The audio encoding device **20** may also invoke the spatio-temporal interpolation unit **50**. The spatio-temporal interpolation unit **50** may perform spatio-temporal interpolation with respect to the reordered transformed HOA coefficients **33'/35'** to obtain the interpolated foreground signals **49'** (which may also be referred to as the "interpolated nFG signals **49**") and the remaining foreground directional information **53** (which may also be referred to as the "V[k] vectors **53**") (**116**). The audio encoding device **20** may then invoke the coefficient reduction unit **46**. The coefficient reduction unit **46** may perform coefficient reduction with respect to the remaining foreground V[k] vectors **53** based on the background channel information **43** to obtain reduced foreground directional information **55** (which may also be referred to as the reduced foreground V[k] vectors **55**) (**118**).

The audio encoding device **20** may then invoke the quantization unit **52** to compress, in the manner described above, the reduced foreground V[k] vectors **55** and generate coded foreground V[k] vectors **57** (**120**).

The audio encoding device **20** may also invoke the psychoacoustic audio coder unit **40**. The psychoacoustic audio coder unit **40** may psychoacoustic code each vector of the energy compensated ambient HOA coefficients **47'** and the interpolated nFG signals **49'** to generate encoded ambient HOA coefficients **59** and encoded nFG signals **61**. The audio encoding device may then invoke the bitstream generation unit **42**. The bitstream generation unit **42** may generate the bitstream **21** based on the coded foreground directional information **57**, the coded ambient HOA coefficients **59**, the coded nFG signals **61** and the background channel information **43**.

FIG. **6** is a flowchart illustrating exemplary operation of an audio decoding device, such as the audio decoding device **24** shown in FIG. **4**, in performing various aspects of the techniques described in this disclosure. Initially, the audio decoding device **24** may receive the bitstream **21** (**130**). Upon receiving the bitstream, the audio decoding device **24** may invoke the extraction unit **72**. Assuming for purposes of discussion that the bitstream **21** indicates that vector-based reconstruction is to be performed, the extraction unit **72** may

parse the bitstream to retrieve the above noted information, passing the information to the vector-based reconstruction unit **92**.

In other words, the extraction unit **72** may extract the coded foreground directional information **57** (which, again, may also be referred to as the coded foreground V[k] vectors **57**), the coded ambient HOA coefficients **59** and the coded foreground signals (which may also be referred to as the coded foreground nFG signals **59** or the coded foreground audio objects **59**) from the bitstream **21** in the manner described above (**132**).

The audio decoding device **24** may further invoke the dequantization unit **74**. The dequantization unit **74** may entropy decode and dequantize the coded foreground directional information **57** to obtain reduced foreground directional information 55_k (**136**). The audio decoding device **24** may also invoke the psychoacoustic decoding unit **80**. The psychoacoustic audio decoding unit **80** may decode the encoded ambient HOA coefficients **59** and the encoded foreground signals **61** to obtain energy compensated ambient HOA coefficients **47'** and the interpolated foreground signals **49'** (**138**). The psychoacoustic decoding unit **80** may pass the energy compensated ambient HOA coefficients **47'** to the fade unit **770** and the nFG signals **49'** to the foreground formulation unit **78**.

The audio decoding device **24** may next invoke the spatio-temporal interpolation unit **76**. The spatio-temporal interpolation unit **76** may receive the reordered foreground directional information $55_k'$ and perform the spatio-temporal interpolation with respect to the reduced foreground directional information $55_k/55_{k-1}$ to generate the interpolated foreground directional information $55_k''$ (**140**). The spatio-temporal interpolation unit **76** may forward the interpolated foreground V[k] vectors $55_k''$ to the fade unit **770**.

The audio decoding device **24** may invoke the fade unit **770**. The fade unit **770** may receive or otherwise obtain syntax elements (e.g., from the extraction unit **72**) indicative of when the energy compensated ambient HOA coefficients **47'** are in transition (e.g., the AmbCoeffTransition syntax element). The fade unit **770** may, based on the transition syntax elements and the maintained transition state information, fade-in or fade-out the energy compensated ambient HOA coefficients **47'** outputting adjusted ambient HOA coefficients **47''** to the HOA coefficient formulation unit **82**. The fade unit **770** may also, based on the syntax elements and the maintained transition state information, and fade-out or fade-in the corresponding one or more elements of the interpolated foreground V[k] vectors $55_k''$ outputting the adjusted foreground V[k] vectors $55_k'''$ to the foreground formulation unit **78** (**142**).

The audio decoding device **24** may invoke the foreground formulation unit **78**. The foreground formulation unit **78** may perform matrix multiplication the nFG signals **49'** by the adjusted foreground directional information $55_k'''$ to obtain the foreground HOA coefficients **65** (**144**). The audio decoding device **24** may also invoke the HOA coefficient formulation unit **82**. The HOA coefficient formulation unit **82** may add the foreground HOA coefficients **65** to adjusted ambient HOA coefficients **47''** so as to obtain the HOA coefficients **11'** (**146**).

FIG. **7** is a diagram illustrating a portion **250** of the bitstream **21** shown in the example of FIGS. **2-4**. The portion **250** shown in the example of FIG. **7**, may be referred to as an HOAConfig portion **250** of the bitstream **21** and includes an HOAOrder field, a MinAmbHoaOrder field, the direction info field **253**, the CodedSpatialInterpolationTime field **254**, the SpatialInterpolationMethod field **255**, the CodedVVecLength field **256** and the gain info field **257**. As shown in the example of FIG. **7**, the CodedSpatialInterpolationTime field **254** may comprise a three bit field, the SpatialInterpolationMethod field **255** may comprise a one bit field, and the CodedVVecLength field **256** may comprise two bit field.

The portion **250** also includes a SingleLayer field **240** and a FrameLengthFactor field **242**. The SingleLayer field **240** may represent one or more bits indicative of whether multiple layers are used to represent the coded version of the HOA coefficients or whether a single layer is used to represent the coded version of the HOA coefficients. The FramelengthFactor field **242** represents one or more bits indicative of a frame length factor, which is discussed in more detail below with respect to FIG. **12**.

FIG. **8** is a diagram illustrating example frames **249**S and **249**T specified in accordance with various aspects of the techniques described in this disclosure. In the example of FIG. **8**, frames **249**S and **249**T each include four transport channels **275**A-**275**D. The transport channel **275**A includes a header bits indicative of ChannelSideInfoData **154**A and HOAGainCorrectionData. The transport channel **275**A also includes a payload bits indicative of VVectorData **156**A. The transport channel **275**B includes a header bits indicative of ChannelSideInfoData **154**B and HOAGainCorrectionData. The transport channel **275**B also includes a payload bits indicative of VVectorData **156**B. The transport channels **275**C and **275**D are not utilized for the frame **249**S. The frame **275**T is substantially similar to the frame **249**S in terms of transport channels **275**A-**275**D.

FIG. **9** is a diagram illustrating example frames for one or more channels of at least one bitstream in accordance with techniques described herein. The bitstream **450** includes frames **810**A-**810**H that may each include one or more channels. The bitstream **450** may be one example of the bitstream **21** shown in the example of FIG. **9**. In the example of FIG. **9**, the audio decoding device **24** maintains state information, updating the state information to determine how to decode the current frame k. The audio decoding device **24** may utilize state information from config **814**, and frames **810**B-**810**D.

In other words, the audio encoding device **20** may include, within the bitstream generation unit **42** for example, the state machine **402** that maintains state information for encoding each of frames **810**A-**810**E in that the bitstream generation unit **42** may specify syntax elements for each of frames **810**A-**810**E based on the state machine **402**.

The audio decoding device **24** may likewise include, within the bitstream extraction unit **72** for example, a similar state machine **402** that outputs syntax elements (some of which are not explicitly specified in the bitstream **21**) based on the state machine **402**. The state machine **402** of the audio decoding device **24** may operate in a manner similar to that of the state machine **402** of the audio encoding device **20**. As such, the state machine **402** of the audio decoding device **24** may maintain state information, updating the state information based on the config **814** and, in the example of FIG. **9**, the decoding of the frames **810**B-**810**D. Based on the state information, the bitstream extraction unit **72** may extract the frame **810**E based on the state information maintained by the state machine **402**. The state information may provide a number of implicit syntax elements that the audio encoding device **20** may utilize when decoding the various transport channels of the frame **810**E.

FIG. **10** illustrates a representation of techniques for obtaining a spatio-temporal interpolation as described herein. The spatio-temporal interpolation unit **50** of the audio encoding device **20** shown in the example of FIG. **3**

may perform the spatio-temporal interpolation described below in more detail. The spatio-temporal interpolation may include obtaining higher-resolution spatial components in both the spatial and time dimensions. The spatial components may be based on an orthogonal decomposition of a multi-dimensional signal comprised of higher-order ambisonic (HOA) coefficients (or, as HOA coefficients may also be referred, "spherical harmonic coefficients").

In the illustrated graph, vectors V_1 and V_2 represent corresponding vectors of two different spatial components of a multi-dimensional signal. The spatial components may be obtained by a block-wise decomposition of the multi-dimensional signal. In some examples, the spatial components result from performing a block-wise form of SVD with respect to each block (which may refer to a frame) of higher-order ambisonics (HOA) audio data (where this ambisonics audio data includes blocks, samples or any other form of multi-channel audio data). A variable M may be used to denote the length of an audio frame in samples.

Accordingly, V_1 and V_2 may represent corresponding vectors of the foreground V[k] vectors $\mathbf{51}_k$ and the foreground V[k−1] vectors $\mathbf{51}_{k-1}$ for sequential blocks of the HOA coefficients **11**. V_1 may, for instance, represent a first vector of the foreground V[k−1] vectors $\mathbf{51}_{k-1}$ for a first frame (k−1), while V_2 may represent a first vector of a foreground V[k] vectors $\mathbf{51}_k$ for a second and subsequent frame (k). V_1 and V_2 may represent a spatial component for a single audio object included in the multi-dimensional signal.

Interpolated vectors V_x for each x is obtained by weighting V_1 and V_2 according to a number of time segments or "time samples", x, for a temporal component of the multi-dimensional signal to which the interpolated vectors V_x may be applied to smooth the temporal (and, hence, in some cases the spatial) component. Assuming an SVD composition, as described above, smoothing the nFG signals **49** may be obtained by doing a vector division of each time sample vector (e.g., a sample of the HOA coefficients **11**) with the corresponding interpolated V_x. That is, $US[n]=HOA[n]*V_x[n]^{-1}$, where this represents a row vector multiplied by a column vector, thus producing a scalar element for US. $V_x[n]^{-1}$ may be obtained as a pseudoinverse of $V_x[n]$.

With respect to the weighting of V_1 and V_2, V_1 is weighted proportionally lower along the time dimension due to the V_2 occurring subsequent in time to V_1. That is, although the foreground V[k−1] vectors $\mathbf{51}_{k-1}$ are spatial components of the decomposition, temporally sequential foreground V[k] vectors $\mathbf{51}_k$ represent different values of the spatial component over time. Accordingly, the weight of V_1 diminishes while the weight of V_2 grows as x increases along t. Here, d_1 and d_2 represent weights.

FIG. **11** is a block diagram illustrating artificial US matrices, US_1 and US_2, for sequential SVD blocks for a multi-dimensional signal according to techniques described herein. Interpolated V-vectors may be applied to the row vectors of the artificial US matrices to recover the original multi-dimensional signal. More specifically, the spatio-temporal interpolation unit **50** may multiply the pseudo-inverse of the interpolated foreground V[k] vectors **53** to the result of multiplying nFG signals **49** by the foreground V[k] vectors $\mathbf{51}_k$ (which may be denoted as foreground HOA coefficients) to obtain K/2 interpolated samples, which may be used in place of the K/2 samples of the nFG signals as the first K/2 samples as shown in the example of FIG. **11** of the U_2 matrix.

FIG. **12** is a block diagram illustrating decomposition of subsequent frames of a higher-order ambisonics (HOA) signal using Singular Value Decomposition and smoothing of the spatio-temporal components according to techniques described in this disclosure. Frame n−1 and frame n (which may also be denoted as frame n and frame n+1) represent subsequent frames in time, with each frame comprising 1024 time segments and having HOA order of 4, giving $(4+1)^2=25$ coefficients. US-matrices that are artificially smoothed U-matrices at frame n−1 and frame n may be obtained by application of interpolated V-vectors as illustrated. Each gray row or column vectors represents one audio object.

Compute HOA Representation of Active Vector Based Signals

The instantaneous CVECk is created by taking each of the vector based signals represented in XVECk and multiplying it with its corresponding (dequantized) spatial vector, VVECk. Each VVECk is represented in MVECk. Thus, for an order N HOA signal, and M vector based signals, there will be M vector based signals, each of which will have dimension given by the frame-length, P. These signals can thus be represented as: XVECkmn, n=0, . . . P−1; m=0, . . . M−1. Correspondingly, there will be M spatial vectors, VVECk of dimension $(N+1)^2$. These can be represented as MVECkml, l=0, . . . , $(N+1)^{2-1}$; m=0, . . . , M−1. The HOA representation for each vector based signal, CVECkm, is a matrix vector multiplication given by:

$$CVECkm=(XVECkm(MVECkm)T)T$$

which, produces a matrix of $(N+1)^2$ by P. The complete HOA representation is given by summing the contribution of each vector based signal as follows:

$$CVECk=m=0M-1CVECk[m]$$

Spatio-Temporal Interpolation of V-Vectors

However, in order to maintain smooth spatio-temporal continuity, the above computation is only carried out for part of the frame-length, P-B. The firstB samples of a HOA matrix, are instead carried out by using an interpolated set of MVECkml, m=0, . . . , M−1; l=0, . . . , $(N+1)^2$, derived from the current MVECkm and previous values MVECk−1m. This results in a higher time density spatial vector as we derive a vector for each time sample, p, as follows:

$$MVECkmp=pB-1MVECkm+B-1-pB-1MVECk-1m,$$
$$p=0, \ldots ,B-1.$$

For each time sample, p, a new HOA vector of (N+1)2 dimension is computed as:

$$CVECkp=(XVECkmp)MVECkmp,\ p=0, \ldots ,B-1$$

These, firstB samples are augmented with the P-B samples of the previous section to result in the complete HOA representation, CVECkm, of the mth vector based signal.

At the decoder (e.g., the audio decoding device **24** shown in the example of FIG. **5**), for certain distinct, foreground, or Vector-based-predominant sound, the V-vector from the previous frame and the V-vector from the current frame may be interpolated using linear (or non-linear) interpolation to produce a higher-resolution (in time) interpolated V-vector over a particular time segment. The spatio temporal interpolation unit **76** may perform this interpolation, where the spatio-temporal interpolation unit **76** may then multiple the US vector in the current frame with the higher-resolution interpolated V-vector to produce the HOA matrix over that particular time segment.

Alternatively, the spatio-temporal interpolation unit **76** may multiply the US vector with the V-vector of the current frame to create a first HOA matrix. The decoder may additionally multiply the US vector with the V-vector from

the previous frame to create a second HOA matrix. The spatio-temporal interpolation unit **76** may then apply linear (or non-linear) interpolation to the first and second HOA matrices over a particular time segment. The output of this interpolation may match that of the multiplication of the US vector with an interpolated V-vector, provided common input matrices/vectors.

In some examples, the size of the time segment for which interpolation is to be performed may vary as a function of the frame length. In other words, the audio encoding device **20** may be configured to operate with respect to a certain frame length or configurable to operate with respect to a number of different frame lengths. Example frame lengths that the audio encoding device **20** may support include 768, 1024, 2048 and 4096. The different frame lengths may result in different sets of possible time segment lengths (where a time segment may be specified in terms of the number of samples). The following table specifies different sets of possible time segments lengths that vary as a function of the frame length (which may be denoted by the variable L).

	CodedSpatialInterpolationTime							
L	0	1	2	3	4	5	6	7
768	0	32	64	128	256	384	512	768
1024	0	64	128	256	384	512	768	1024
2048	0	128	256	512	768	1024	1536	2048
4096	0	256	512	1024	1536	2048	3072	4096

In the foregoing table, the syntax element "CodedSpatialInterpolationTime" represents one or more bits indicative of a spatial interpolation time. The variable L denotes the frame length, as noted above. For a frame length of 768, the possible time segment lengths are defined by, in this example, the set of 0, 32, 64, 128, 256, 384, 512 and 768. The one value used for the current frame is specified by the value of the CodedSpatialInterpolationTime syntax element, where a value of zero indicates a time segment length of 0, a value of one indicates a time segment length of 32 and so on. For a frame length of 1024, the possible time segment lengths are defined, in this example, by the set of 0, 64, 128, 256, 384, 512, 768, and 1024. The one value used for the current frame is specified by the value of the CodedSpatialInterpolationTime syntax element, where a value of zero indicates a time segment length of 0, a value of one indicates a time segment length of 64 and so on. For a frame length of 2048, the possible time segment lengths are defined by the set of 0, 128, 256, 512, 768, 1024, 1536, and 2048. The one value used for the current frame is specified by the value of the CodedSpatialInterpolationTime syntax element, where a value of zero indicates a time segment length of 0, a value of one indicates a time segment length of 128 and so on. For a frame length of 4096, the possible time segment lengths are defined by, in this example, the set of 0, 256, 512, 1024, 1536, 2048, 3072, and 4096. The one value used for the current frame is specified by the value of the CodedSpatialInterpolationTime syntax element, where a value of zero indicates a time segment length of 0, a value of one indicates a time segment length of 256 and so on.

The spatio-temporal interpolation unit **50** of the audio encoding device **20** may perform the interpolation with respect to a number of different time segments selected form the corresponding set identified by the frame length, L. The spatio-temporal interpolation unit **50** may select the time segment that sufficiently smoothes the transition across the frame boundary (e.g., in terms of a signal to noise ratio) and that requires the least number of samples (given that interpolation may be a relatively expensive operation in terms of power, complexity, operations, etc.).

The spatio-temporal interpolation unit **50** may obtain the frame length, L, in any number of different ways. In some examples, the audio encoding device **20** is configured with a pre-set frame rate (which may be hard coded or, in other words, configured statically or manually configured as part of configuring the audio encoding device **20** to encode the HOA coefficients **11**). In some examples, the audio encoding device **20** may specify the frame length based on a core coder frame length of the psychoacoustic audio coder unit **40**. More information regarding the core coder frame length may be found with regard to discussions of a "coreCoderFrameLength" in ISO/IEC 23003-3:2012, entitled "Information technology—MPEG audio technologies—Part 3: Unified speech and audio coding."

When determined based on the core coder frame length, the audio encoding device **20** may reference the following table:

TABLE

	FrameLengthFactor definition			
Core Coder Frame Length	FrameLengthFactor			
	00	01	10	11
	$C_{FrameLength}$			
768	1	1	1	Reserved
1024	1	1	1	
2048	1	1/2	1/2	
4096	1	1/2	1/4	

In the foregoing table, the audio encoding device **20** may set one or more bits (denoted by the syntax element "FrameLengthFactor") indicating the factor by which to multiple the core coder frame length, which is specified in the first column of the above table. The audio encoding device **20** may select one of the frame length factors of 1, ½ and ¼ based on various coding criteria or may select one of the factors based on attempts at coding frames at each of the various factors. The audio encoding device **20** may for example determine that the core coder frame length is 4096 and select a frame length factor of 1, ½ or ¼. The audio encoding device **20** may signal the frame length factor in an HOAConfig portion of the bitstream **21** (as shown above with respect to the example of FIG. **7**), where a value of 00 (binary) indicates a frame length factor of 1, a value of 01 (binary) indicates a frame length factor of ½, and a value of 10 (binary) indicates a frame length factor of ¼. The audio encoding device **20** may also determine the frame length L as the core coder frame length multiplied by the frame length factor (e.g., 1, ½, or ¼).

In this respect, the audio encoding device **20** may obtain a time segment based, at least in part, on one or more bits indicative of a frame length (L) and one or more bits indicative of a spatio-temporal interpolation time (e.g., the codedSpatioInterpolationTime syntax element). The audio encoding device **20** may also obtain decomposed interpolated spherical harmonic coefficients for the time segment by, at least in part, performing an interpolation with respect to the first decomposition of the first plurality of spherical harmonic coefficients and the second decomposition of the second plurality of spherical harmonic coefficients.

The audio decoding device **24** may perform substantially similar operations to those described above with respect to

the audio encoding device **20**. In particular, the spatio-temporal interpolation unit **76** of the audio decoding device **24** may obtain the frame length as a function of the one or more bits indicative of the frame length factor (e.g., the frameLengthFactor syntax element) and the core coder frame length (which may also be specified in the bitstream **21** by the psychoacoustic audio encoding unit **40**). The spatio-temporal interpolation unit **76** may also obtain the one or more bits indicative of the spatio-temporal interpolation time (e.g., the CodedSpatialInterpolationTime syntax element). The spatio-temporal interpolation unit **76** may perform a lookup in the table noted above using the frame length L and the codedSpatialInterpolationTim syntax element as keys to identifying the time segment length. The audio decoding device **24** may then perform the interpolation in the manner described above for the obtained time segment.

In this respect, the audio decoding device **24** may obtain a time segment based, at least in part, on one or more bits indicative of a frame length (L) and one or more bits indicative of a spatio-temporal interpolation time (e.g., the codedSpatioInterpolationTime syntax element). The audio decoding device **24** may also obtain decomposed interpolated spherical harmonic coefficients for the time segment by, at least in part, performing an interpolation with respect to the first decomposition of the first plurality of spherical harmonic coefficients and the second decomposition of the second plurality of spherical harmonic coefficients.

FIG. **13** is a diagram illustrating one or more an audio encoder and an audio decoder configured to perform one or more techniques described in this disclosure. As discussed above, SVD may be utilized as a basis for an HOA-signal compression system. In some examples, an HOA signal H may be decomposed into USV' (' is a transpose of a matrix). In some examples, the first few rows of US and V matrices may be defined as background signals (e.g., ambient signals), and the first few columns of US and V matrices may be defined as foreground signals. In some examples, the background and foreground signals may be crossfaded in a similar way. However, crossfading the background and foreground signals in a similar may result in redundant calculations being performed. To reduce the calculations performed and to improve other aspects of the system, this disclosure describes a new crossfading algorithm for the background signal.

In some systems, the US matrix and the V matrix are separately crossfaded into the US_C matrix (e.g., the crossfaded US matrix) and the V_C matrix (e.g., the crossfaded V matrix), respectively. Then, the crossfaded HOA signal H_C may be reconstructed as US_C*V_C'. In accordance with one or more techniques of this disclosure, the original HOA signal H may be reconstructed as USV' (e.g., prior to crossfading). Crossfading may then be performed in the HOA domain as described throughout this disclosure.

As noted above, the length (or in other words, the number of samples) of the frame may vary (e.g., as a function of the core coder frame length). The difference in frame length along with the different sets of the spatio-temporal interpolation times may impact crossfading as described above. In general, the spatio-temporal interpolation time identified by the CodedSpatialInterpolationTime syntax element and the frame length L may specify the number of samples to be crossfaded. As shown in the example of FIG. **13**, the size of the U matrix is (L+SpatialInterpolationTime)*25, where the SpatialInterpolationTime variable denotes the spatial interpolation time as obtained as a function of the CodedSpatialInterpolationTime syntax element and L using the table discussed above with respect to FIG. **12**. An example value for the SpatialInterpolationTime may be 256 when L equals 1024 and the value of the CodedSpatialInterpolationTime syntax element equals three. Another example value for the SpatialInterpolationTime, which will be used for purposes of illustration below, may be 512 when L equals 2048 and the value of the CodedSpatialInterpolationTime syntax element equals three. Under this illustrative example, the L+SpatialInterpolationTime equals 2048+512 or 2560.

In any event, the background HOA coefficients are of dimension 2560*4 in this example. The crossfade therefore occurs between the SptailInterpolationTime number of samples (e.g., 512 samples) of the previous frame and the first SptailInterpolationTime number of samples (e.g., 512 samples) of the current frame. The output is therefore L samples, which are AAC or USAC coded. Accordingly, the SpatialInterpolationTime used for spatio-temporally interpolating V-vectors may also identify the number of samples over which crossfading is performed. In this way, the crossfading duration may be impacted by one or more bits indicative of the FrameLength and one or more bits indicative of a spatio-temporal interpolation time.

Moreover, the energy compensation unit **38** may perform the energy compensation to generate the ambient HOA coefficients **47'** by applying a windowing function to the $V_{BG}[k]$ vectors $\mathbf{35}_{BG}$ to generate the energy compensated $V_{BG}[k]$ vectors $\mathbf{35}_{BG}'$. The windowing function may comprise a windowing function having a length equal to the frame length L. In this respect, the energy compensation unit **38** may use the same frame length L for the energy compensation obtained, at least in part, on the one or more bits indicative of the frame length factor (e.g., the FrameLengthFactor syntax element).

the mixer unit **70** of the crossfade unit **66** may combine (e.g., crossfade between) the first portion **73** of the energy compensated ambient HOA coefficients **47'** of the current frame and the third portion **76** of the energy compensated ambient HOA coefficients **47'** of the previous frame to generate intermediate crossfaded energy compensated ambient HOA coefficients **77**. The mixer unit **70** may output the generated intermediate crossfaded energy compensated ambient HOA coefficients **77** to the framing unit **71**. As the mixer unit **70** utilizes the third portion **76** of the energy compensated ambient HOA coefficients **47'** of the previous frame, in this example, it can be assumed that the crossfade unit **66** was in operation prior to processing the current frame. As such, as opposed to separately crossfading the US matrix of the current frame with the US matrix of the previous frame and the V matrix of the current frame with the V matrix of the previous frame, the mixer unit **70** may crossfade in the energy compensated domain. In this way, techniques according to the disclosure may reduce the computational load, power consumption, and/or complexity of the crossfade unit **66**.

The foregoing techniques may be performed with respect to any number of different contexts and audio ecosystems. A number of example contexts are described below, although the techniques should be limited to the example contexts. One example audio ecosystem may include audio content, movie studios, music studios, gaming audio studios, channel based audio content, coding engines, game audio stems, game audio coding/rendering engines, and delivery systems.

The movie studios, the music studios, and the gaming audio studios may receive audio content. In some examples, the audio content may represent the output of an acquisition. The movie studios may output channel based audio content (e.g., in 2.0, 5.1, and 7.1) such as by using a digital audio

workstation (DAW). The music studios may output channel based audio content (e.g., in 2.0, and 5.1) such as by using a DAW. In either case, the coding engines may receive and encode the channel based audio content based one or more codecs (e.g., AAC, AC3, Dolby True HD, Dolby Digital Plus, and DTS Master Audio) for output by the delivery systems. The gaming audio studios may output one or more game audio stems, such as by using a DAW. The game audio coding/rendering engines may code and or render the audio stems into channel based audio content for output by the delivery systems. Another example context in which the techniques may be performed comprises an audio ecosystem that may include broadcast recording audio objects, professional audio systems, consumer on-device capture, HOA audio format, on-device rendering, consumer audio, TV, and accessories, and car audio systems.

The broadcast recording audio objects, the professional audio systems, and the consumer on-device capture may all code their output using HOA audio format. In this way, the audio content may be coded using the HOA audio format into a single representation that may be played back using the on-device rendering, the consumer audio, TV, and accessories, and the car audio systems. In other words, the single representation of the audio content may be played back at a generic audio playback system (i.e., as opposed to requiring a particular configuration such as 5.1, 7.1, etc.), such as audio playback system **16**.

Other examples of context in which the techniques may be performed include an audio ecosystem that may include acquisition elements, and playback elements. The acquisition elements may include wired and/or wireless acquisition devices (e.g., Eigen microphones), on-device surround sound capture, and mobile devices (e.g., smartphones and tablets). In some examples, wired and/or wireless acquisition devices may be coupled to mobile device via wired and/or wireless communication channel(s).

In accordance with one or more techniques of this disclosure, the mobile device may be used to acquire a soundfield. For instance, the mobile device may acquire a soundfield via the wired and/or wireless acquisition devices and/or the on-device surround sound capture (e.g., a plurality of microphones integrated into the mobile device). The mobile device may then code the acquired soundfield into the HOA coefficients for playback by one or more of the playback elements. For instance, a user of the mobile device may record (acquire a soundfield of) a live event (e.g., a meeting, a conference, a play, a concert, etc.), and code the recording into HOA coefficients.

The mobile device may also utilize one or more of the playback elements to playback the HOA coded soundfield. For instance, the mobile device may decode the HOA coded soundfield and output a signal to one or more of the playback elements that causes the one or more of the playback elements to recreate the soundfield. As one example, the mobile device may utilize the wireless and/or wireless communication channels to output the signal to one or more speakers (e.g., speaker arrays, sound bars, etc.). As another example, the mobile device may utilize docking solutions to output the signal to one or more docking stations and/or one or more docked speakers (e.g., sound systems in smart cars and/or homes). As another example, the mobile device may utilize headphone rendering to output the signal to a set of headphones, e.g., to create realistic binaural sound.

In some examples, a particular mobile device may both acquire a 3D soundfield and playback the same 3D soundfield at a later time. In some examples, the mobile device may acquire a 3D soundfield, encode the 3D soundfield into HOA, and transmit the encoded 3D soundfield to one or more other devices (e.g., other mobile devices and/or other non-mobile devices) for playback.

Yet another context in which the techniques may be performed includes an audio ecosystem that may include audio content, game studios, coded audio content, rendering engines, and delivery systems. In some examples, the game studios may include one or more DAWs which may support editing of HOA signals. For instance, the one or more DAWs may include HOA plugins and/or tools which may be configured to operate with (e.g., work with) one or more game audio systems. In some examples, the game studios may output new stem formats that support HOA. In any case, the game studios may output coded audio content to the rendering engines which may render a soundfield for playback by the delivery systems.

The techniques may also be performed with respect to exemplary audio acquisition devices. For example, the techniques may be performed with respect to an Eigen microphone which may include a plurality of microphones that are collectively configured to record a 3D soundfield. In some examples, the plurality of microphones of Eigen microphone may be located on the surface of a substantially spherical ball with a radius of approximately 4 cm. In some examples, the audio encoding device **20** may be integrated into the Eigen microphone so as to output a bitstream **21** directly from the microphone.

Another exemplary audio acquisition context may include a production truck which may be configured to receive a signal from one or more microphones, such as one or more Eigen microphones. The production truck may also include an audio encoder, such as audio encoder **20** of FIG. **3**.

The mobile device may also, in some instances, include a plurality of microphones that are collectively configured to record a 3D soundfield. In other words, the plurality of microphone may have X, Y, Z diversity. In some examples, the mobile device may include a microphone which may be rotated to provide X, Y, Z diversity with respect to one or more other microphones of the mobile device. The mobile device may also include an audio encoder, such as audio encoder **20** of FIG. **3**.

A ruggedized video capture device may further be configured to record a 3D soundfield. In some examples, the ruggedized video capture device may be attached to a helmet of a user engaged in an activity. For instance, the ruggedized video capture device may be attached to a helmet of a user whitewater rafting. In this way, the ruggedized video capture device may capture a 3D soundfield that represents the action all around the user (e.g., water crashing behind the user, another rafter speaking in front of the user, etc. . . .).

The techniques may also be performed with respect to an accessory enhanced mobile device, which may be configured to record a 3D soundfield. In some examples, the mobile device may be similar to the mobile devices discussed above, with the addition of one or more accessories. For instance, an Eigen microphone may be attached to the above noted mobile device to form an accessory enhanced mobile device. In this way, the accessory enhanced mobile device may capture a higher quality version of the 3D soundfield than just using sound capture components integral to the accessory enhanced mobile device.

Example audio playback devices that may perform various aspects of the techniques described in this disclosure are further discussed below. In accordance with one or more techniques of this disclosure, speakers and/or sound bars may be arranged in any arbitrary configuration while still playing back a 3D soundfield. Moreover, in some examples,

headphone playback devices may be coupled to a decoder **24** via either a wired or a wireless connection. In accordance with one or more techniques of this disclosure, a single generic representation of a soundfield may be utilized to render the soundfield on any combination of the speakers, the sound bars, and the headphone playback devices.

A number of different example audio playback environments may also be suitable for performing various aspects of the techniques described in this disclosure. For instance, a 5.1 speaker playback environment, a 2.0 (e.g., stereo) speaker playback environment, a 9.1 speaker playback environment with full height front loudspeakers, a 22.2 speaker playback environment, a 16.0 speaker playback environment, an automotive speaker playback environment, and a mobile device with ear bud playback environment may be suitable environments for performing various aspects of the techniques described in this disclosure.

In accordance with one or more techniques of this disclosure, a single generic representation of a soundfield may be utilized to render the soundfield on any of the foregoing playback environments. Additionally, the techniques of this disclosure enable a rendered to render a soundfield from a generic representation for playback on the playback environments other than that described above. For instance, if design considerations prohibit proper placement of speakers according to a 7.1 speaker playback environment (e.g., if it is not possible to place a right surround speaker), the techniques of this disclosure enable a render to compensate with the other 6 speakers such that playback may be achieved on a 6.1 speaker playback environment.

Moreover, a user may watch a sports game while wearing headphones. In accordance with one or more techniques of this disclosure, the 3D soundfield of the sports game may be acquired (e.g., one or more Eigen microphones may be placed in and/or around the baseball stadium), HOA coefficients corresponding to the 3D soundfield may be obtained and transmitted to a decoder, the decoder may reconstruct the 3D soundfield based on the HOA coefficients and output the reconstructed 3D soundfield to a renderer, the renderer may obtain an indication as to the type of playback environment (e.g., headphones), and render the reconstructed 3D soundfield into signals that cause the headphones to output a representation of the 3D soundfield of the sports game.

In each of the various instances described above, it should be understood that the audio encoding device **20** may perform a method or otherwise comprise means to perform each step of the method for which the audio encoding device **20** is configured to perform In some instances, the means may comprise one or more processors. In some instances, the one or more processors may represent a special purpose processor configured by way of instructions stored to a non-transitory computer-readable storage medium. In other words, various aspects of the techniques in each of the sets of encoding examples may provide for a non-transitory computer-readable storage medium having stored thereon instructions that, when executed, cause the one or more processors to perform the method for which the audio encoding device **20** has been configured to perform.

In one or more examples, the functions described may be implemented in hardware, software, firmware, or any combination thereof. If implemented in software, the functions may be stored on or transmitted over as one or more instructions or code on a computer-readable medium and executed by a hardware-based processing unit. Computer-readable media may include computer-readable storage media, which corresponds to a tangible medium such as data storage media. Data storage media may be any available media that can be accessed by one or more computers or one or more processors to retrieve instructions, code and/or data structures for implementation of the techniques described in this disclosure. A computer program product may include a computer-readable medium.

Likewise, in each of the various instances described above, it should be understood that the audio decoding device **24** may perform a method or otherwise comprise means to perform each step of the method for which the audio decoding device **24** is configured to perform. In some instances, the means may comprise one or more processors. In some instances, the one or more processors may represent a special purpose processor configured by way of instructions stored to a non-transitory computer-readable storage medium. In other words, various aspects of the techniques in each of the sets of encoding examples may provide for a non-transitory computer-readable storage medium having stored thereon instructions that, when executed, cause the one or more processors to perform the method for which the audio decoding device **24** has been configured to perform.

By way of example, and not limitation, such computer-readable storage media can comprise RAM, ROM, EEPROM, CD-ROM or other optical disk storage, magnetic disk storage, or other magnetic storage devices, flash memory, or any other medium that can be used to store desired program code in the form of instructions or data structures and that can be accessed by a computer. It should be understood, however, that computer-readable storage media and data storage media do not include connections, carrier waves, signals, or other transitory media, but are instead directed to non-transitory, tangible storage media. Disk and disc, as used herein, includes compact disc (CD), laser disc, optical disc, digital versatile disc (DVD), floppy disk and Blu-ray disc, where disks usually reproduce data magnetically, while discs reproduce data optically with lasers. Combinations of the above should also be included within the scope of computer-readable media.

Instructions may be executed by one or more processors, such as one or more digital signal processors (DSPs), general purpose microprocessors, application specific integrated circuits (ASICs), field programmable logic arrays (FPGAs), or other equivalent integrated or discrete logic circuitry. Accordingly, the term "processor," as used herein may refer to any of the foregoing structure or any other structure suitable for implementation of the techniques described herein. In addition, in some aspects, the functionality described herein may be provided within dedicated hardware and/or software modules configured for encoding and decoding, or incorporated in a combined codec. Also, the techniques could be fully implemented in one or more circuits or logic elements.

The techniques of this disclosure may be implemented in a wide variety of devices or apparatuses, including a wireless handset, an integrated circuit (IC) or a set of ICs (e.g., a chip set). Various components, modules, or units are described in this disclosure to emphasize functional aspects of devices configured to perform the disclosed techniques, but do not necessarily require realization by different hardware units. Rather, as described above, various units may be combined in a codec hardware unit or provided by a collection of interoperative hardware units, including one or more processors as described above, in conjunction with suitable software and/or firmware.

Various aspects of the techniques have been described. These and other aspects of the techniques are within the scope of the following claims.

The invention claimed is:

1. A method comprising:

obtaining, by an audio encoder, a decomposition of spherical harmonic coefficients (SHCs) that correspond to a first set of ambient SHCs, the decomposition including a first set of vectors representing spatial characteristics of an ambient sound field and a second set of vectors representing temporal and energy characteristics of the ambient sound field;

performing, by the audio encoder, energy compensation with respect to the first set of vectors to obtain a set of energy compensated vectors;

multiplying, by the audio encoder, the set of energy compensated vectors by the second set of vectors to obtain a first set of energy compensated ambient SHCs;

crossfading, by the audio encoder, between the first set of energy compensated ambient spherical harmonic coefficients (SHCs) and a second set of energy compensated ambient SHCs to obtain a first set of crossfaded energy compensated ambient SHCs.

2. The method of claim 1,

wherein the first set of SHCs include SHCs corresponding to basis functions having an order greater than one, and

wherein the second set of SHCs include SHCs corresponding to basis functions having an order greater than one.

3. The method of claim 1, wherein performing the energy compensation comprises performing the energy compensation using a windowing function obtained as a function, at least in part, of one or more bits indicative of a frame length.

4. The method of claim 1,

wherein the first set of energy compensated ambient SHCs correspond to a current frame, and

wherein the second set of energy compensated ambient SHCs correspond to a previous frame.

5. The method of claim 1, wherein crossfading comprises modifying a portion of the first set of energy compensated ambient SHCs based on a portion of the second set of energy compensated ambient SHCs.

6. An audio decoding device comprising:

a memory configured to store a first set of vectors representing spatial characteristics of a foreground sound field, a second set of vectors representing temporal and energy characteristics of the foreground sound field, a first set of energy compensated ambient spherical harmonic coefficients (SHCs), and a second set of energy compensated ambient SHCs, wherein the first set of energy compensated ambient SHCs describe a first ambient sound field and the second set of energy compensated ambient SHCs describe a second ambient sound field, and

one or more processors, coupled to the memory, configured to:

crossfade between the first set of energy compensated ambient SHCs and the second set of energy compensated ambient SHCs to obtain a first set of crossfaded energy compensated ambient SHCs; and

render, based on the first set of vectors, the second set of vectors, and the first set of crossfaded energy compensated ambient SHCs, one or more speaker feeds.

7. The audio decoding device of claim 6,

wherein the first set of SHCs include SHCs corresponding to basis functions having an order greater than one, and

wherein the second set of SHCs include SHCs corresponding to basis functions having an order greater than one.

8. The audio decoding device of claim 6,

wherein the first set of energy compensated ambient SHCs correspond to a current frame, and

wherein the second set of energy compensated ambient SHCs correspond to a previous frame.

9. The audio decoding device of claim 6, wherein the one or more processors are configured to crossfade by at least modifying a portion of the first set of energy compensated ambient SHCs based on a portion of the second set of energy compensated ambient SHCs.

10. The audio decoding device of claim 6, further comprising a speaker configured to reproduce, based on the speaker feeds, a sound field.

11. An audio encoding device comprising:

one or more processors configured to:

obtain a decomposition of spherical harmonic coefficients (SHCs) that correspond to a first set of ambient SHCs, the decomposition including a first set of vectors representing spatial characteristics of an ambient sound field and a second set of vectors representing temporal and energy characteristics of the ambient sound field;

perform energy compensation with respect to the first set of vectors to obtain a set of energy compensated vectors;

multiply the set of energy compensated vectors by the second set of vectors to obtain a first set of energy compensated ambient SHCs;

a memory, coupled to the one or more processors, configured to store the first set of energy compensated ambient spherical harmonic coefficients (SHCs) and a second set of energy compensated ambient SHCs, and

wherein the one or more processors are further configured to crossfade between the first set of energy compensated ambient SHCs and the second set of energy compensated ambient SHCs to obtain a first set of crossfaded energy compensated ambient SHCs.

12. The audio encoding device of claim 11,

wherein the first set of SHCs include SHCs corresponding to basis functions having an order greater than one, and

wherein the second set of SHCs include SHCs corresponding to basis functions having an order greater than one.

13. The audio encoding device of claim 11, wherein the one or more processors are configured to perform the energy compensation using a windowing function obtained as a function, at least in part, of one or more bits indicative of a frame length.

14. The audio encoding device of claim 11,

wherein the first set of energy compensated ambient SHCs correspond to a current frame, and

wherein the second set of energy compensated ambient SHCs correspond to a previous frame.

15. The audio encoding device of claim 11, wherein the one or more processors are configured to crossfade by at least modifying a portion of the first set of energy compensated ambient SHCs based on a portion of the second set of energy compensated ambient SHCs.

16. The audio encoding device of claim 11, further comprising a microphone configured to capture audio data indicative of the first and second sets of ambient SHCs.

17. The method of claim 1, wherein the device is the audio encoder, the method further comprises capturing, by a microphone coupled to the audio encoder, audio data representative of a first set of ambient SHCs and a second set of ambient SHCs.

18. The method of claim 1, wherein the device is the audio decoder, the method further comprises:

rendering one or more loudspeaker feeds based on the first set of crossfaded energy compensated ambient SHCs; and

reproducing, by one or more loudspeakers coupled to the audio decoder and based on the one or more loudspeaker feeds, a crossfaded ambient soundfield represented by the first set of crossfaded energy compensated ambient SHCs.

19. The audio decoding device of claim **6**,

wherein the audio decoding device further comprises one or more speakers coupled to the one or more processors and configured to reproduce, based on the one or more speaker feeds, a sound field.

20. A method comprising:

obtain, by an audio decoder, a first set of vectors representing spatial characteristics of a foreground sound field and a second set of vectors representing temporal and energy characteristics of the foreground sound field,

obtain, by the audio decoder, a first set of energy compensated ambient spherical harmonic coefficients (SHCs) and a second set of energy compensated ambient SHCs, wherein the first set of energy compensated ambient SHCs describe a first ambient sound field and the second set of energy compensated ambient SHCs describe a second ambient sound field, and

crossfading, by the audio decoder, between the first set of energy compensated ambient SHCs and the second set of energy compensated ambient SHCs to obtain a first set of crossfaded energy compensated ambient SHCs; and

render, by the audio decoder and based on the first set of vectors, the second set of vectors, and the first set of crossfaded energy compensated ambient SHCs, one or more speaker feeds.

21. The method of claim **20**,

wherein the first set of SHCs include SHCs corresponding to basis functions having an order greater than one, and

wherein the second set of SHCs include SHCs corresponding to basis functions having an order greater than one.

22. The method of claim **20**,

wherein the first set of energy compensated ambient SHCs correspond to a current frame, and

wherein the second set of energy compensated ambient SHCs correspond to a previous frame.

23. The method of claim **20**, wherein crossfading between the first set of energy compensated ambient SHCs and the second set of energy compensated SHCs comprises crossfading by at least modifying a portion of the first set of energy compensated ambient SHCs based on a portion of the second set of energy compensated ambient SHCs.

24. The method of claim **20**, further comprising reproducing, by one or more speakers and based on the one or more speaker feeds, a sound field.

25. The method of claim **20**, wherein obtaining the first set of energy compensated SHCs and the second set of energy compensated SHCs comprises obtaining a bitstream that includes a representation of the crossfaded energy compensated ambient SHCs and a representation of crossfaded foreground SHCs that correspond to the crossfaded energy compensated ambient SHCs.

26. The method of claim **20**, wherein the first set of vectors and the second set of vectors represent crossfaded foreground SHCs, and wherein obtaining the first set of vectors and the second set of vectors comprises obtaining a bitstream that includes a representation of the crossfaded foreground SHCs.

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